SCOOP 4+

Stereo audio codec for real time audio transmission

User manual

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1. General

The SCOOP 4+ codec allows the bi-directional transmission of one or two audio signals with bit rate reduction, over digital leased lines, ISDN lines or IP protocol networks. The codec is available with the following main product versions:

- SCOOP 4+ DUO LL/IP, with digital leased line interfaces and an Ethernet interface for IP transmission;
- SCOOP 4+ TRIO LL/IP/ISDN 2B, with digital leased line interfaces, Ethernet interface and an ISDN interface;
- SCOOP 4+ TRIO LL/IP/ISDN 4B, with digital leased line interfaces, Ethernet interface and two ISDN interfaces.

The following table shows the main features of the product. Functions marked with ● in this table are available as options. Functions marked with □ are only available in the versions equipped with ISDN interface(s).

One outstanding feature of AAS codecs in ISDN mode is the 5A System®: on receiving an incoming ISDN call, the unit can automatically detect the coding algorithm and parameters of the calling codec, and then adjust itself in a compatible configuration so that the connection succeeds regardless of the initial configuration and that of the remote unit.

In IP mode, the codec features the same ease of operation thanks to the use of the SIP and SDP protocols.

The standard operation mode is the "single codec" mode, where the unit can be connected to a remote codec using any one of the listed coding algorithms.

In the dual 7 kHz codec mode (available for leased line transmission), the equipment is equivalent to two independent mono codecs, each running G722 over a 64 kbit/s leased lines.

Such a dual codec mode is also available for ISDN transmission; in this case the equipment is equivalent to two independent mono codecs, each running G711 or G722 over one B channel of the ISDN interface.

® 5AS = Aeta Audio Advanced Automatic Adjustment System

Characteristics		Optional
Operation modes		
Single wide band codec		
Dual 7 kHz codec (LL mode or ISDN mode)		
IP transmission interface		
Ethernet interface, 10BaseT / 100BaseT; TCP/IP and UDP/IP protocols		
Audio transmission in unicast mode: SIP signalling protocol, SDP protocol		
Audio transmission in multicast mode		
Leased line transmission interfaces		
Two X24/X21/V11/V35 interfaces;		
64, 128, 192, 256 or 384 kbit/s over one line, or 2x64 kbit/s over two lines		
ISDN transmission interface		
One or two S0 interfaces (U interfaces available for North America)		
5AS automatic setting for incoming calls		□
Audio coding algorithms	(audio modes)	
G711 (standard telephone)	Mono	
G722 SRT, H221, H242	Mono	
CELP 7 kHz (IP mode only)	Mono	
MPEG Audio Layer II	Mono, Stereo, Dual mono, Joint stereo	
MPEG AAC-LC	Mono, Stereo	●
MPEG HE-AAC, HE-AAC v2	Mono, Stereo	●
4 sub-band ADPCM (low latency)	M, S	
TDAC (ISDN mode only)	M	□ ●
Available bit rates (depending on coding algorithm):		
Leased line transmission: 64 to 384 kbit/s over one line, or 2x64 kbit/s over two lines		
IP transmission: 16 kbit/s to 256 kbit/s		
ISDN: 64 to 256 kbit/s transmitted via one or two interfaces (1 to 4 B channels)		□
Audio interfaces		
Two analogue inputs and two analogue outputs with adjustable level		
Digital audio input and output, AES/EBU format		
Auxiliary functions		
Data channel, 300 to 9600 bauds		
Relay transmission: 2 isolated inputs and outputs		
Audio coordination channel		●
Control and supervision		
Keyboard and LCD display on front panel		
99 programmable set-up/dial memories		
Remote control serial port		
Ethernet remote control interface		
Embedded html server		
Isolated control and status loops		

Table 1 – Main features of the SCOOP 4+

2. Functions

The following synoptic diagram shows the basic functions of the equipment.

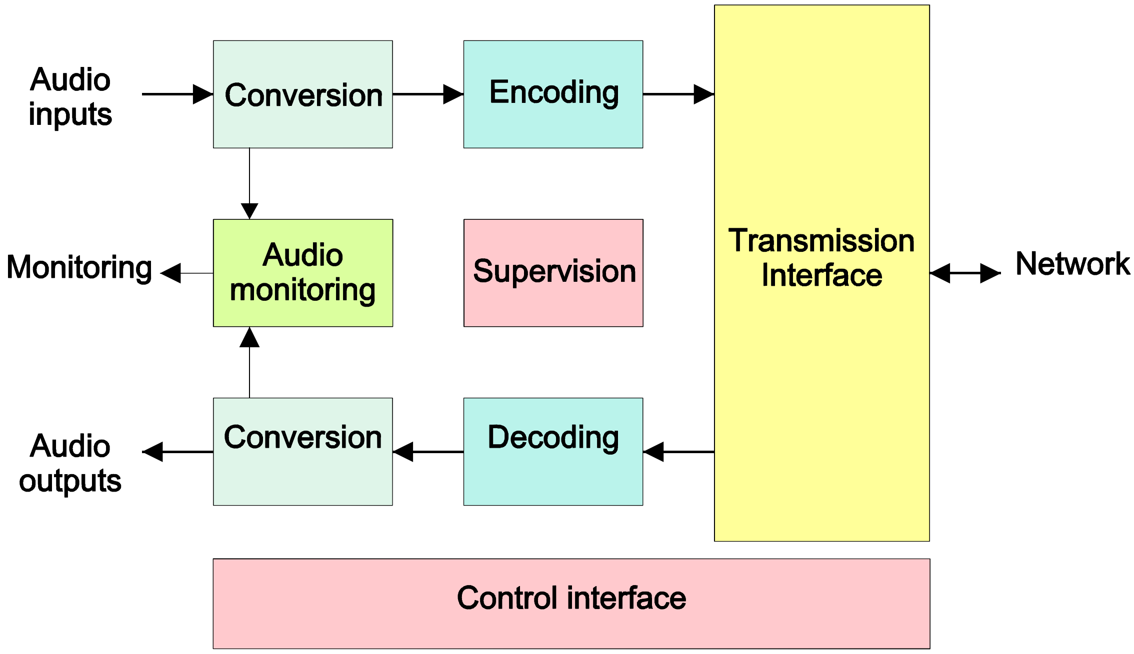


Figure 1 - Functional diagram of equipment

The audio signals to be transmitted are converted (when needed) to digital format, then the encoding function reduces the bit rate, using a selectable algorithm; the resulting bit flow is sent to one of the available transmission interfaces: permanent link data interfaces (X21/X24/V35), ISDN interfaces (S0 or U0), or an Ethernet interface.

The transmission interface functional block also extracts compressed data coming from the network and sends them to a decoding block that reproduces uncompressed audio data. Last, the audio signals are output to both digital and analogue outputs.

2.1. Conversion of audio signals

The analogue inputs and outputs are balanced, and the input and output gains are adjustable. The sampling frequency of the analogue ⇔ digital converters depend on the operating mode.

The equipment also provides digital audio inputs/outputs in AES/EBU format. The input to the encoder is selectable between the digital audio input and the analogue stereo input. The output from the decoder is sent both to the digital output and the analogue stereo output. The digital audio interfaces are usually locked to the digital audio input (“genlock” mode), but alternatively they can be synchronised to the internal clock reference of the codec.

Having the digital samples from the audio interfaces (analogue or digital), sample rate conversion is fulfilled whenever needed to get audio data at the coding frequency F_c which is, depending on the coding type, 16, 24, 32 or 48 kHz.

2.2. Encoding and decoding

In the normal single codec mode, the codec readily includes a wide range of coding algorithms. First, one can select among algorithms compliant with ISO and ITU-T recommendations:

- G711 (IP or ISDN mode only);
- ITU-T G722 (mono at 64 kbit/s);
- MPEG Audio Layer II at 48, 32, 24 or 16 kHz, with programmable channel mode and bit rate ;
- MPEG AAC-LC at 48 or 32 kHz, programmable channel mode and bit rate; available as an option, for IP and ISDN modes;
- MPEG HE-AAC and HE-AAC v2 at 48 or 32 kHz, programmable channel mode and bit rate; available as an option[1] for IP and ISDN modes;

Besides, other algorithms are available, that are so-called "proprietary" because they do not comply with enforced standards:

- CELP, running in mono at a net 24 kbit/s bit rate, and providing a 7 kHz bandwidth (only used in IP mode);
- 4SB ADPCM, running either in mono at a 128 kbit/s bit rate, or in stereo at 256 kbit/s; the bandwidth with this algorithm is 15 kHz.
- TDAC mono, running at 64 kbit/s, with a 15 kHz bandwidth; available as an option in ISDN mode.

In the dual 7 kHz codec mode, each codec (or audio channel) can be encoded using the following algorithms:

- G711; which is the standard coding for voice transmission on the ISDN (this algorithm is not available in LL mode);
- ITU-T G722, running in mono at a 64 kbit/s rate.

The following describes some important features of the various available algorithms and protocols.

2.2.1. Notes about G711

G711 is the standard coding used for voice transmission on public telephone networks. This algorithm is typically used for links over IP networks with IP telephones or VoIP gateways. Via ISDN, G711 is used for links with telephones or hybrid devices.

G711 is available only for IP or ISDN transmission, not over the leased line interfaces.

2.2.2. Notes about G722

With G722 coding, three synchronisation modes are available:

- "Statistical recovery" byte synchronisation method (alias SRT) ;
- H221 synchronisation; in this case, 1.6 kbit/s from the compressed data are used for this;
- H221 synchronisation and H242 protocol. This is only available for the ISDN mode.

H221 synchronisation is highly recommended when possible, as it features higher reliability and faster recovery time, while degradation (because of the bit rate used for framing) is minimal.

H242 protocol, the most flexible mode, is recommended by the ITU-T, and is included in J52. However, the mode with H221 synchronisation but without H242 protocol can be useful for compatibility with old generation codecs which did not use this protocol.

No specific synchronisation is needed for the IP mode.

[1] Not available for first generation units

2.2.3. Notes about MPEG coding and J52

The ITU-T J52 recommendation was defined in order to allow the interoperability of multimedia terminals over the ISDN[1], using common coding standards. It includes the following features:

- Framing as per ITU-T H221 recommendation, ensuring byte synchronisation and interchannel synchronisation when more than one 64 kbit/s B channel is required for the desired bit rate ;
- Interoperation procedures according to ITU-T H242 recommendation ;
- In the case of MPEG encoding, optional protection against transmission errors (Reed-Solomon error correction codes). Although J52 does not apply to leased line connections, this error protection technique is also available for leased line transmission with the SCOOP 4+.

Details about MPEG and J52 can be found in the annexes (refer to 6.1, Complements on the algorithms and protocols used).

It must be noted that, thanks to the interoperation protocol, J52 codecs, when setting up a link, can negotiate automatically and agree on a configuration that is compatible with the capability of both units (regarding bit rate, channel mode, etc.). In this way, when the units differ in their capability (or make), the resulting configuration may be different from expected beforehand, but in most cases the link will work and audio will be transmitted.

As another useful consequence, this also gives users more tolerance to mistakes when configuring the units on the two sides of the transmission links, as the codecs will adapt automatically even with differences in the initial settings of the two units.

2.2.4. MPEG coding for leased line or IP transmission

J52 is only applicable to ISDN transmission, and no inverse multiplexing is needed for leased line transmission neither for IP transmission, because a single data stream is transmitted.

For these reasons, only one MPEG format is defined for non-ISDN transmission; there is no distinction in these modes between J52-compliant or non compliant format.

2.2.5. Notes about TDAC

As an option, the codec can also include the TDAC algorithm. TDAC is for Time Domain Aliasing Cancellation ; this is a transform coding based on an MDCT (Modified Discrete Cosine Transform), encoding a 15 kHz bandwidth mono signal at a 64 kbit/s bit rate.

Some specific product versions also include “asymmetric” modes:

- G722/TDAC : G722 encoding, TDAC decoding, running both in mono at 64 kbit/s ;
- TDAC/G722 : TDAC encoding, G722 decoding (with SRT), running both in mono at 64 kbit/s ; this mode is symmetric to the previous one.

[1] J52 is only relevant for ISDN connections

2.2.6. Symmetric or asymmetric codec modes

The codec allows two communication modes:

Symmetric communication: in this mode, the encoder and decoder both use the same coding algorithm with the same configuration (channel mode, etc.). In this case, the communication is strictly symmetric full-duplex, with exactly the same coding configuration used in both directions (local to remote and remote to local). This is usually required when using proprietary algorithms.

Asymmetric communication: this mode is used for applications requiring different coding configurations in the two directions. The J52 protocol allows such mode. To give some examples, it is possible to transmit MPEG Layer II in one direction and G722 in the other one, or MPEG stereo in one direction and MPEG mono in the other one, etc.

Specific product versions also allow asymmetric modes wherein one direction is G722 coded while the other one is TDAC coded. Such mode is useful e.g. in order to get a low delay return path encoded in G722 while the send path is encoded with higher quality but a higher delay.

2.2.7. 5A System®

Setting an ISDN connection is often difficult, at least because of the numerous coding parameters to be set. Moreover, with most proprietary algorithms, it is mandatory for the two devices to have exactly the same settings, otherwise the connection will fail, and sometimes it is not easy to find out the reason.

5A stands for Aeta Audio Advanced Automatic Adjustment. This system makes it easier to set an ISDN connection, because the codec, on receiving a call, automatically adjusts itself, following the calling party algorithm and parameters.

When the 5A System is enabled on the unit and a call is received, the unit first detects the coding algorithm used by the calling codec, and also senses its parameters: audio mode (mono, stereo…), sampling rate, bit rate, inverse multiplexing protocol, etc. Then the unit can decode the compressed audio from the remote unit. In addition, the unit will use these same settings for encoding and sending audio to the remote unit, so that the remote unit can also decode the outgoing audio programme. The whole process just takes a few seconds. Of course, all compatible coding configurations can be detected automatically by the 5A System.

Note that the 5A system is only active for ISDN connections.

2.2.8. SIP protocol and SDP

The SIP protocol is a signalling protocol, used for IP connections, which allows the SCOOP4+ to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Details about the SIP protocols can be found in the annex (refer to 6.2, Overview of the SIP protocol).

One significant advantage is the inclusion of SDP, a protocol which allows the connecting devices to automatically negotiate and agree on the coding profile to use. Thanks to this system, it is not necessary to set the units in the same way before setting up a connection. Moreover, the calling party need not know how the remote unit is configured before initiating a link.

- ✓ *Note: the SIP protocol does not mandatorily imply the use of a server. Codecs can set up point-to-point links using this protocol, and benefit from some advantages of this protocol.*

2.3. Transmission interfaces

The codec includes an Ethernet interface for IP protocol networks, interfaces for transmission over leased lines, and one or two ISDN interfaces are available in some versions.

2.3.1. Ethernet/IP interface

The IP interface is a 10BaseT/100BaseT Ethernet interface allowing transmission of the audio programmes in a wide range of possible bit rates.

IP unicast mode

The most classical transmission mode is unicast: audio connection with one distant device, generally bidirectional. This mode can be used on all types of networks links, LAN or WAN, including links via Internet. The SCOOP 4+ implements the SIP protocol, which allows it to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Links can be set up in two ways:

- "Peer to peer" connection between two compatible units
- Use of a SIP proxy server to set up the link, or a SIP PBX

Details about the SIP protocols can be found in the annex (refer to 6.2, Overview of the SIP protocol).

The audio coding algorithm can be selected depending on the required quality and the available network bandwidth. The following algorithms are currently available:

Codec	Bit rate (coding)	Bit rate (total)[1]	Audio bandwidth	Typical use, main features
G711	64 kbit/s	86 kbit/s	3 kHz	Voice, telephony Compatible with IP phones
CELP	24 kbit/s	28,5 kbit/s	7 kHz	Suitable for high quality speech; Low network bandwidth consumption
G722	64 kbit/s	86 kbit/s	7 kHz	High quality speech
MPEG Layer II	64 or 128 kbit/s	83 to 147 kbit/s	up to 20 kHz	Highest quality, suitable for speech and music
MPEG AAC-LC	16 à 192 kbit/s	30 to 213 kbit/s	up to 20 kHz	Low bit rate, suitable for speech and music
MPEG HE-AAC and HE-AAC v2	16 to 128 kbit/s	23 to 139 kbit/s	up to 20 kHz	Very low bit rate, suitable for speech and music
4SB ADPCM	64 or 128 kbit/s	173 or 301 kbit/s	15 kHz	Low latency, suitable for speech and music

IP multicast mode

The multicast mode allows an encoder device to transmit an audio programme to several decoders by sending a single encoded stream to a multicast group address. The link is unidirectional by nature. This mode can be used on a local area network, and on larger private networks that can manage the multicast mode. On the other hand, Internet cannot support this routing mode.

[1] Informative value; higher than the "net" encoded audio bit rate because of the protocol overhead

In this mode, SCOOP 4+ uses the RTP protocol to manage the audio stream, like in the unicast mode, but the SIP protocol is not applicable here; instead a proprietary signalling system is used. As the link is unidirectional, the unit has to be set either as a “sender” in order to encode and transmit the audio stream to the selected group address, or as a “receiver” to receive and decode such stream coming from a “sender” device.

The audio coding algorithm can be selected with just the same capability as for the unicast mode described above.

Remote control via IP

In addition, the Ethernet interface can be used for configuring or remote controlling the unit via a TCP/IP connection.

- An embedded web server allows configuring some parameters using a web browser, via port 80 (default port for the HTTP protocol);
- TCP port 6000 can be used for a “command line” control mode, typically used by codec supervision software.

2.3.2. Leased line interfaces

For transmission over leased lines, the codec includes two X24/V11 ports which can run at 64 kbit/s, 128 kbit/s, 192 kbit/s, 256 kbit/s and 384 kbit/s bit rates.

With most coding modes, only one X24/V11 port is used. In the “2*64” dual mono G722 mode, the two ports provide two independent interfaces; the equipment is similar to two mono codecs.

When transmitting in the “leased line” mode, the codec synchronises onto the network clock provided by the X24/V11 interface. In the specific “2*64” mode where the two ports are used, the codec initially synchronises on port #1, but it changes the synchronisation port in case of a fault.

If no valid clock is available on the X24/V11 interfaces, the system folds back to an internal clock.

2.3.3. ISDN Interface

On the ISDN side, the codec includes one or two BRI interfaces (S0 or U0 depending on equipment version), allowing transmission over one to four 64 kbit/s B channels. Thus, the total available bit rate ranges from 64 to 128 kbit/s.

The codec synchronises itself onto the ISDN network clock when a link is active.

2.3.4. Use of a backup connection to secure a permanent link

When a fixed link (LL) is used, it is possible to use another network access (IP or ISDN) in order to set a temporary backup link in case the nominal LL link fails. The unit will then switch to a backup mode (ISDN or IP depending on the selected backup arrangement), and provide the audio transmission via the backup network access. More precisely, on one end of the link the codec will switch to the backup mode and “call” its counterpart via the ISDN or and IP network. On the other end the unit will switch to the backup mode when it receives the call on its ISDN or IP interface. The operating mode and configuration for this backup feature are detailed further in following sections (4.10, Setting up a backup link).

2.4. Supervision and control interface

These functional modules fulfil the control and supervision of the equipment (configuration, communication management, status monitoring), thanks to a keyboard, an alphanumeric display, LED indicators, and remote control interfaces.

The remote control is possible either via a serial data port or through the Ethernet interface with a TCP/IP connection.

In addition, the equipment features a "Loop control" function: call set up and release can be remote controlled with current loops and relays, instead of using for this the keyboard and/or the remote control port.

In order to allow easy and quick programming of the codec for specific operational configurations, the equipment features configuration memories (or "profiles"). When recalling a profile, the codec is directly reconfigured with parameters that were stored beforehand in this profile by the operator, and/or the stored number is dialled by the unit.

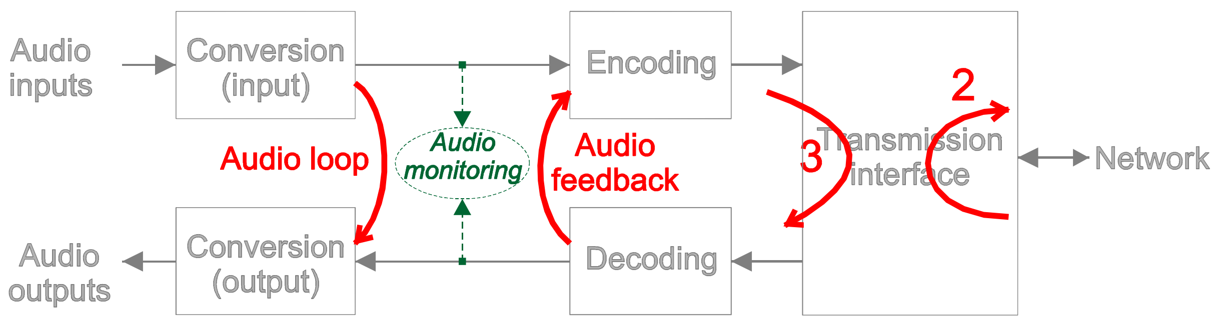
Besides configuring the equipment operating mode, this module monitors its status (detection of alarm conditions). On detecting operation or transmission faults, the equipment switches on indicators and relay contacts. Two alarm classes are defined:

- "Internal" alarm ; corresponds to a major fault internal to the equipment ;
- "External" alarm ; corresponds to a fault whose origin is deemed external to the equipment (for example, transmission fault);

For maintenance purposes, some test loops can be activated:

- "Audio loop": uncompressed audio data are looped from the input of the encoder to the input of the output conversion functional block. This loop redirects the audio input to the audio outputs;
- "Loop 3", or "Codec" loop: compressed audio data are looped just before the network interface ;
- "Loop 2", or "Network" loop: this loop sends the received data back to the network ; for the remote codec, the effect is the same as a loop 3 when the transmission works correctly ;
- "Audio feedback" loop (audio output to audio input) ; this allows the codec to send back to the remote codec the signal it receives, after decoding and re-encoding.

The following drawing schematically shows the test loops:



2.5. Audio monitoring

This function enables the monitoring of the audio input (before encoding) and the audio output (after decoding the received signal), and provides:

- A display of the signal level both at the encoder input and the decoder output ;
- A test output on a stereo headphone jack, monitoring either the encoder or decoder audio signals.

✓ *Note: as the audio output is monitored immediately after decoding, this monitoring position is not sensitive to the possible activation of the audio test loop (see above diagram), contrarily to the physical audio outputs (both analog and digital).*

2.6. Auxiliary functions

2.6.1. Data channel

This function is not available in IP transmission mode.

In leased line mode or ISDN mode, a bi-directional data channel can be transmitted along with the compressed audio signals, by reserving a fraction of the transmitted bit rate. The equipment includes a serial asynchronous port for this purpose. The data are transparently transmitted end-to-end; hardware signalling is not available.

This function is only available when the main audio programme is encoded in G722 H242 (ISDN mode), MPEG (excluding the proprietary ISDN mode) or ADPCM.

The interface speed is programmable at 300, 1200, 2400, 4800 or 9600 bauds. However, the actual transmission capacity depends on the coding algorithm, as indicated by the table hereunder.

Coding type	Possible transmission rate (bit/s)				
	300	1200	2400	4800	9600
G722 (H221/H242)					
MPEG Audio					
4SB ADPCM					

Table 1 – Capacity of data channel depending on type of coding

2.6.2. Relay transmission

When this function is activated, the codec transmits to the remote unit the status of two isolated current loops. The remote unit then opens or closes relay contacts according to the transmitted status. Conversely, as the function is bi-directional, the codec activates its two relays (“dry” isolated contacts) depending on the status of the two current loops on the remote unit.

In IP transmission mode, this function is available whatever the selected coding.

In LL or ISDN mode, this function is only available when the main audio programme is encoded in G722 H242 (ISDN mode), MPEG (excluding the proprietary ISDN mode) or ADPCM.

When using MPEG coding, relay transmission can be activated along with other auxiliary functions. For G722 or ADPCM, relay transmission is activated in place of the data channel.

A typical application is the transmission of an “on air” signal; the contact closure may be used for e.g. switching on a lamp or starting other devices.

2.6.3. Coordination channel

This function is currently only available in leased line transmission mode.

This function is available as an option. It enables the transmission of an auxiliary audio channel (or coordination or “order-wire” channel), along with the compressed audio, by reserving 8 kbit/s from the transmitted bit rate. This channel uses a compression algorithm of CELP-HLTP type and provides a “voice grade” channel (300-3400 Hz pass-band).

This function is only available when the main audio programme is encoded in G722 H242 (ISDN mode), MPEG (excluding the proprietary ISDN mode) or ADPCM.

With G722 or ADPCM, the coordination channel cannot be used along with other auxiliary functions (i.e. data channel or relay transmission).

When using MPEG coding, all three auxiliary functions can be activated at the same time. Note that relay transmission and the coordination channel are only compatible with AAS products, as these functions are not covered by independent standards.

3. Operation

3.1. General principles

The equipment control and supervision (configuration, status monitoring) is possible in two ways:

- “Local” mode: front panel keyboard and display, status indicators ;
- “Remote control” mode, thanks to an asynchronous serial port or the Ethernet interface.

As a general rule, the configuration parameters are saved in non-volatile memory, and restored when the unit is powered-on.

Local mode operation is described in detail in chapter 4 (Detailed operating mode).

Thanks to the remote control mode, the codec can be operated from a computer with supervision software. The supervision station is a PC computer running Windows, equipped with the Scoop4Man configuration and monitoring software. This software gives full access to the codec functions (configuration, audio link management and status monitoring) with a graphical interface.

Scoop 4+ can also be controlled by the optional software TeleScoop, which can control the other AAS codecs of the Hifiscoop, Scoop 3 and Hifiscoop 3 ranges. Details about this supervision software can be found in the documentation and user manual of the TeleScoop software.

In addition to this, some parameters related to the Ethernet/IP interface and transmission can be set by using an embedded HTML server; these are described further in 3.6, “Use of the embedded html server”.

For controlling connections in ISDN or IP mode, it is also possible to use the “Loop control” function. When this special connection mode is selected, one can trigger a call by activating an input current loop (optically isolated), and release the line by de-activating this loop. In such case, an outgoing connection is established or released only by this way, and no more from the front panel or the remote control interface (however, all other parameters are still controlled from these interfaces as in the normal mode).

If “loop control” is not activated, it is always possible to use the loop to release a running connection (a pulse on the loop will release the line).

Besides, whatever the connection mode (normal or loop control), a “dry loop” is closed when a connection is active.

The loop control interfaces are described in 3.2.2 and 5.1.13.

The SCOOP 4+ can be remote controlled by third-party codec management software and systems. Please consult us for more information on the available offer in this field.

3.2. Physical description of the equipment

The SCOOP 4+ codec is housed in a 19 inches chassis of 1U height (44 mm or 1.75”). It includes a universal mains power supply, or optionally it can be powered from a 12V DC source.

3.2.1. Front panel

All the elements needed for local control are located on the front panel (see picture on page 16 below). This panel can be roughly divided in three areas:

On the left-hand side, one can find an LCD and the basic navigation and dialling keys. The central area of the panel includes several status LEDs and a keypad for the entry of dialling numbers and/or text data. The right hand side groups audio monitoring elements.

- ✓ ***The “Esc” key is also used to power the unit on and off***:
  When the unit is in standby (the blue LED besides the Esc key is on), hold the key depressed for at least 3 seconds to switch the unit on;
  When the unit is in operation, press the key down for more than 3 seconds to switch it off.

In addition to this “soft” switch, the unit automatically switches on when AC power is applied to its power socket.

LCD and basic control keypad

This part is used for configuration and call set up; details cane be found in chapter 4, dealing with the operating mode.

The 2x20 character alphanumeric display is surrounded by the following keys, (*from left to right*):

Key	Function
“**Hang up**”	Release a link in IP or ISDN transmission mode
“**Unhook**”	Start a link or accept an incoming call (in IP or ISDN transmission mode)
Navigation keys	Menu-dependent keys; used to scroll options and/or select an option in a menu. The bottom line on the LCD shows the function of each key.
“**OK**” key	Confirm a selection or enter data
“**Esc**” / ⏻ key	*Short pressure*: Escape to higher menu level; *Long pressure*: Switch on or switch off the unit[1]

The blue LED besides the Esc/Power key is off in operation, but lights on when the unit is in standby.

[1] Note: the standby mode can be disabled by internal configuration; in such case a long pressure has no effect!

Status LED indicators

The LEDs have the following meaning (*from left to right*):

Marking	*Color*	Function
⏻	*blue*	On when unit is in standby
Line 1	*Green*	On when interface n°1 is active / connected
Line 2	*Green*	On when interface n°2 is active / connected
Dec 1	*Green*	On when the decoder is synchronised on “line 1”
Dec 2	*Green*	On when the decoder is synchronised on “line 2”
INFO 1	*Amber*	Displays the status of the received “relay info” n°1
INFO 2	*Amber*	Displays the status of the received “relay info” n°2
Test	*Red*	On whenever a test loop is active
Alarm / Ext	*Red*	On in case of an external alarm
Alarm / Int	*Red*	On in case of an alarm with internal cause

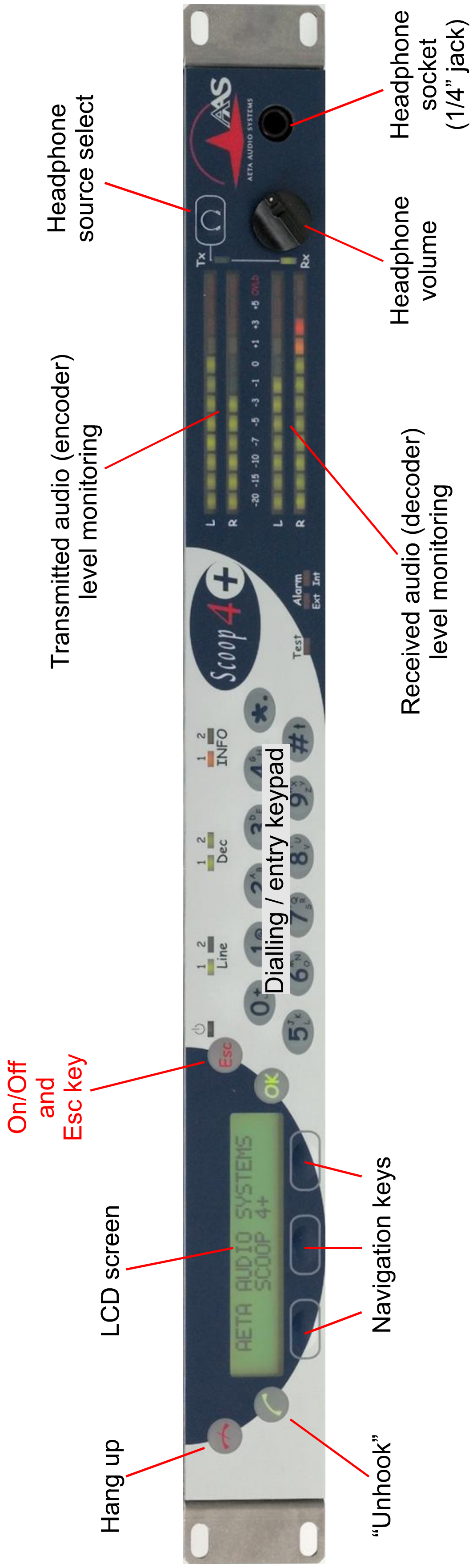


Figure 2 - Front panel of SCOOP 4+

Audio monitoring

Two pairs of LED bargraphs display the level of the audio signals, both on the transmission and reception directions. The top bargraphs display the level of the audio channels on the transmitter (encoder), while the bottom bargraphs display the level of the received channels (decoder side). The 0 dB mark is a reference level that can be adjusted (*relatively to digital full scale; the reference level can be set in the menu, SETUP / Audio / Level Meter / HEADROOM)*. The “OVLD” LED at the right end of each bar shows when the signal reaches maximum digital level (or clipping level), regardless of the reference level setting.

✓ *“OVLD” always reacts to <u>absolute</u> full scale level, while the bargraph level indication depends on the reference level setting*

The “HEADROOM” setting in the menu defines (in dB) the difference between the *maximum level* (or digital full scale) and the *reference level*, for which 0 dB is displayed on the level meters. Here are some examples:

- If the headroom is set at 0 dB, then the maximum displayed level is 0 dB; note that OVLD will light on whenever the signal reaches this level (or exceeds it on the analogue input).
- If the headroom is set at 10 dB, then “0 dB” is displayed when the signal is 10 dB below maximum level, or -10 dBFS. The display in such case can reach up to +5 dB. OVLD lights up when the signal reaches maximum level (but not before!).

The audio signals can also be monitored with a headphone connected on the front panel (1/4” or 6.35 mm stereo jack). The headphone volume is adjustable thanks to a potentiometer, and the source select key toggles the listening between transmission (**Tx** indicator) or reception (**Rx** indicator).

Actions dealing with this area (connecting or disconnecting the jack, Tx/Rx selection, volume adjustment) never affect the transmitted or received signals.

3.2.2. Rear panel

All connections are done on the rear panel of the codec. The characteristics of the interfaces and layout of the sockets are detailed in chapter 5.1. Characteristics of interfaces.

The following elements are available on the rear panel (refer to following Figure 3 - Rear panel):

Mains power socket

This is an IEC type power socket.

The "12V DC" version of the product includes, in place of this AC power socket, a 4-pin male XLR socket for an external DC source. Refer to details in 5.1.14, Connector for DC input (option).

Audio inputs/outputs

- Analog inputs/outputs: at the input, plug the audio cables into the female XLR sockets. At the output, plug the audio cables into the male XLR sockets. In mono mode, only "A" channel is used.
- Digital inputs/outputs: a digital input (mono or stereo) in AES/EBU format (or SPDIF) can be connected on the female XLR socket, and a digital output in AES/EBU format is available on a male XLR socket.
- It is possible to select which input (analog inputs or digital input) is fed to the encoder for transmission. On the receiving side, the decoded signals are output both on the analog and digital outputs.

X24/V11/V35 interfaces (labelled "X24/V11/V35" and "ALARM + X24/V11")

These sockets are used for the connection to data transmission equipment in the "leased line" mode.

The connectors are 15-point male, Sub-D type. In the standard single codec mode, only one port is used. This is normally the main port "X24/V11/V35", but it is possible to select the other port.

In the dual codec mode, both ports must be used. In this mode, audio channel A is transmitted on the main port "X24/V11/V35", and audio channel B is transmitted on the additional port labelled "ALARM + X24/V11".

Alarm indicator and contacts

This "ALARM + X24/V11" port also includes two "form C" relays, providing isolated contacts, which can signal alarm conditions:

- Internal alarm contact;
- External alarm contact;

A red LED indicator also indicates that an alarm relay is activated. In the factory setup, every alarm cause sets the LED on, but by setting jumpers on the motherboard it is possible to program the indicator to react to only one type (internal or external alarm).

The pin-out of the socket and the detailed characteristics of the alarm relays can be found in chapter 5.1.6: "Alarm + X24/X21" interface (p. 51).

USB socket

This host USB port is currently not used.

Figure 3 - Rear panel

Remote control (Remote)

This 9-pin female sub-D socket is an asynchronous serial interface port, usable for remote controlling the equipment thanks to a control and supervision PC.

Data

This 9-pin female sub-D socket is an asynchronous serial interface port, usable for transmission of a bi-directional data channel (refer to 2.6.1 above, Data channel).

Ethernet interface

This socket is a 100BaseT/10BaseT port, used for audio transmission over IP and/or for remote controlling the unit via a TCP/IP connection (TCP port: 6000). This RJ45 socket is devised for a normal "straight" cable to an Ethernet hub or switch. The two integrated LEDs show the presence and activity of the network (green LED) and the interface mode: half-duplex (yellow LED off) or full-duplex (yellow LED on).

The configuration of the interface is described in 3.5, Initial setup of the Ethernet interface.

"ISDN 1" and "ISDN 2" sockets

These RJ45 sockets allow the connection to the ISDN, for the product versions which include this capability. Their layout is standard. The sockets must be used according to their number, i.e. #1 must be used if one line only is needed, #1 and #2 if two lines are needed.

"Digital I/O" socket

Reserved for future use.

"AUX" socket

This 25-pin female sub-D socket groups the interfaces for the relay transmission function and the (optional) coordination audio channel.

It also includes loop interfaces for the loop control function, as well as a +5 V power supply that can be used to provide current for the loop and relay interfaces.

3.3. Equipment configuration parameters

The parameters may be divided into the following categories:

- Coding configuration parameters, which include audio coding type, coding frequency Fc (and subsequently the nominal bandwidth), audio channel mode and transmission bit rate. Besides, in case of MPEG coding, it is possible to select the error protection mode.
- Configuration of the audio interfaces, including: selection of analogue or digital format for the audio input, maximum level for the analogue inputs and outputs, and synchronisation mode for the AES/EBU interfaces.
- Parameters of the auxiliary functions: possible activation of a data channel, bit rate of this, possible activation of the relay transmission, possible activation of the auxiliary audio channel (if this option is available).
- Parameters of the network access: type of network interface (Ethernet/IP, leased line or ISDN), interface parameters, etc.
- Parameters of the keyboard/display interface (as an example, selection of the language for the display messages).

Chapter 4 (Detailed operating mode) describes these two last categories.

The parameters dealing with the audio interfaces are programmable independently from the others. On the other hand, the auxiliary functions depend on the current transmission mode and coding type.

The following table is a summary, for each coding type, of the allowed values for the various parameters of the coding configuration and auxiliary functions.

Meaning of abbreviations in the table:

- Channel mode : M = Mono, S = Stereo, JS = Joint stereo, DM = Dual Mono
- Coding : H221 = H221 synchronisation, SRT = Statistical Recovery Timing
- 4B = only available on 4B equipment version (two ISDN interfaces)
- FEC : Forward Error Correction = Reed-Solomon error correction

Only MPEG can be configured with all three auxiliary functions (data, auxiliary audio, relays). For other algorithms, each function, when available, can only be used alone. Auxiliary functions are only available for codec 1 when in double codec configuration.

Regarding the auxiliary functions, the table does not apply to the IP mode, in which:

- The data channel and the auxiliary audio are not available;
- The relays are always available, regardless of the coding algorithm used.

✓ *The (optional) MPEG AAC coding, in all its variations, is very flexible regarding the bit rate selection in IP transmission mode. To preserve readability, this table does not give the exhaustive list of all available bit rates. However, for ISDN transmission only 64 and 128 kbit/s are allowed.*

Coding	Channel mode	Sampling rate	Band-width	Bit rate	LL	ISDN	IP	Data channel	Relays	Audio aux.	FEC
G711	M	8 kHz	3,4 kHz	64 kbit/s							
CELP	M	16 kHz	7 kHz	24 kbit/s							
G722 (SRT)	M	16 kHz	7 kHz	64 kbit/s							
G722 H221		16 kHz	7 kHz	64 kbit/s							
G722 H242		16 kHz	7 kHz	64 kbit/s				≤ 4800			
4SB ADPCM	M	32 kHz	15 kHz	128 kbit/s				≤ 4800			
4SB ADPCM	S	32 kHz	15 kHz	256 kbit/s							
MPEG Layer II	M	16 kHz	7 kHz	64 kbit/s				≤ 9600			
		16 kHz	7 kHz	128 kbit/s				≤ 9600			
		24 kHz	10 kHz	64 kbit/s				≤ 9600			
		24 kHz	10 kHz	128 kbit/s				≤ 9600			
		32 kHz	15 kHz	64 kbit/s				≤ 9600			
		32 kHz	15 kHz	128 kbit/s				≤ 9600			
		32 kHz	15 kHz	192 kbit/s		4B		≤ 9600			
		48 kHz	20 kHz	64 kbit/s				≤ 9600			
		48 kHz	20 kHz	128 kbit/s				≤ 9600			
		48 kHz	20 kHz	192 kbit/s		4B		≤ 9600			
MPEG Layer II	S,JS,DM	16 kHz	7 kHz	64 kbit/s				≤ 9600			
		16 kHz	7 kHz	128 kbit/s				≤ 9600			
		24 kHz	10 kHz	64 kbit/s				≤ 9600			
		24 kHz	10 kHz	128 kbit/s				≤ 9600			
		32 kHz	15 kHz	64 kbit/s				≤ 9600			
		32 kHz	15 kHz	128 kbit/s				≤ 9600			
		32 kHz	15 kHz	192 kbit/s		4B		≤ 9600			
		32 kHz	15 kHz	256 kbit/s		4B		≤ 9600			
		32 kHz	15 kHz	384 kbit/s				≤ 9600			
		48 kHz	20 kHz	64 kbit/s				≤ 9600			
		48 kHz	20 kHz	128 kbit/s				≤ 9600			
		48 kHz	20 kHz	192 kbit/s		4B		≤ 9600			
		48 kHz	20 kHz	256 kbit/s		4B		≤ 9600			
		48 kHz	20 kHz	384 kbit/s				≤ 9600			
MPEG AAC-LC	M	32 kHz	15 kHz	16 to 192 kbit/s							
		48 kHz	20 kHz	16 to 192 kbit/s							
	S	32 kHz	15 kHz	16 to 192 kbit/s							
		48 kHz	20 kHz	16 to 192 kbit/s							
MPEG HE-AAC	M	32 kHz	15 kHz	16 to 64 kbit/s							
		48 kHz	20 kHz	16 to 64 kbit/s							
	S	32 kHz	15 kHz	16 to 128 kbit/s							
		48 kHz	20 kHz	16 to 128 kbit/s							
MPEG HE-AAC v2	S	32 kHz	15 kHz	16 to 64 kbit/s							
		48 kHz	20 kHz	16 to 64 kbit/s							
TDAC	M	32 kHz	15 kHz	64 kbit/s				≤ 300			

Table 2 – Possible values for configuration parameters

3.4. Installation and set up

3.4.1. Mounting and connections

Natural convection cools the equipment. Avoid obstructing the openings on the flanges.

To operate the codec, the minimum necessary connections to set up are (see details in the rear panel description):

- Power supply ;
- Audio inputs and outputs (XLR sockets) ;
- Network interface: depending on the networks used, Ethernet interface, ISDN lines and/or X24/V11/V35 interface(s) ;

Whenever needed, the "ALARM + X24/V11" socket must be connected to an external supervision system (alarm relay contacts).

The pin out of the connectors is indicated in chapter 5.1**:** Characteristics of interfaces.

3.4.2. Initial set up

Before the first link, the equipment must be configured according to the desired operation mode (audio input/output format, coding type and parameters, etc.) and the local conditions (network interface parameters…).

For using the keyboard, a password may have to be entered. After factory setting or after total configuration erasure, the password is blank (no password needed). Afterwards, a password can be programmed by the user if one is needed.

For more details about the codec configuration, see chapter 3.3 (Equipment configuration parameters, p. 21) and chapter 4 (Detailed operating mode). The setup of the Ethernet interface is described in 3.5 (Initial setup of the Ethernet interface).

3.4.3. Notes about the use of AES/EBU interfaces

When using digital audio interfaces, it must be decided whether the codec is "***master***" or "slave" regarding audio sampling clock synchronisation. In the first case, the codec derives the sampling clock from the network clock or an internal source, and the device(s) connected to the codec must synchronise to the same clock source.

The most common choice is rather the "slave" mode, to be used when it is not possible (or not desired) to synchronise the external equipment onto the clock of the transmission link or the codec. In this case, the AES/EBU interfaces should be set in the so-called "***genlock***" mode. When in this mode, the codec derives the sampling clock of the digital audio interfaces from its AES input (in other words is "gen-locked" onto the incoming AES signal), and sampling rate conversion (SRC) is used for interfacing to the coding parts.

- ✓ *It is mandatory in such situation to provide the codec input with an AES signal featuring the same sampling frequency as the external equipment, even if the codec is used only as a decoder.*

If this requirement is ignored, the unit will actually fall back to "master" mode. In such situation, clicks in the audio programme might be heard, especially when the resulting sampling rate is very different from that of the external device.

If, on the contrary, it is decided to synchronise the external equipment (at 32, 48 or 96 kHz) onto the transmission clock of the leased line interface, the codec must be configured in "master" mode. In this case, the output is locked onto this clock, and it can be used as a reference to synchronise the equipment connected to the codec output. The digital audio signal at the codec input must then come from a device synchronised by this way.

- ✓ *When you do not use the digital audio interfaces, the "master" or "slave" mode has no effect on the actual operation. However, it is recommended in such case to select the "master" mode to avoid undesired alarms. Otherwise, with the "genlock" setting (which is the default factory setting), an alarm is raised because of the lack of a suitable signal on the AES input. In the "master" mode, the device ignores this error condition.*

3.5. Initial setup of the Ethernet interface

The SCOOP 4+ includes a 100BaseT / 10BaseT Ethernet interface, and the audio transmission can take place over an IP network through this interface. In addition, it is always possible to use the Ethernet interface for remote controlling the unit via a TCP/IP connection (TCP port 6000).

For IP (unicast) audio transmission, SCOOP 4+ uses the SIP protocol, which eases the setting up of a link. The operation is similar to setting a call over the ISDN or PSTN. The transmission can be done in two modes:

- Direct "peer to peer" transmission between two compatible units.
- Use of a SIP proxy server for the call setup

The coding algorithm can be selected depending on the desired quality and audio bandwidth. The algorithms currently available are listed in 2.3.1, Ethernet/IP interface.

If you wish to register the codec on a SIP proxy server, you should configure in the device the data for the SIP "account" on this server. This can be done using the embedded html server; refer for that to 3.6.1 below.

For setting into operation, first connect the Ethernet interface to the network, using CAT5 wiring.

- Connection to 10BaseT or 100BaseT interfaces are both suitable, as the SCOOP 4+ automatically switches to the right 10 Mbit/s or 100 Mbit/s mode.
- "Straightforward" patch cables should be used for a connection to a hub or a switch. Conversely, a "crossed" cable might be needed for special configurations (e.g. a test connection to a PC).

As a very first step, the Ethernet interface must be assigned an IP address, and related parameters. This phase is very simple when a DHCP server is available in the network. The menu to use is reached by TOOLS / Maintenance / Ethernet Setup.

- ✓ *When Ethernet/IP is the current interface for audio transmission, an alternate path in the menu is SETUP / Net / Param / Network Setup.*

3.5.1. DHCP server available

This is the simplest case, because the server will allocate a suitable IP address and give the unit the right settings. Select "DHCP" in the menu (TOOLS / Maintenance / Ethernet Setup). The unit will then automatically find the DHCP server and automatically set the parameters. You can read the IP address (allocated to the unit by the DHCP server) in the "About" menu (TOOLS / Maintenance / About).

Note that, as an additional advantage with DHCP, you do not need to change this setting later, even if you move the SCOOP 4+ to another network, as long as it is still connected to a DHCP server.

3.5.2. “Static” IP configuration

When there is no DHCP server, you have to enter the settings manually, using the menu (TOOLS / Maintenance / Ethernet Setup / Manual / etc.). The IP address must be “available”, i.e. not already assigned to other equipment. Ask support from the network administrator(s) as needed. The following has to be entered:

Parameter	Notes
IP address	Must be unique on the network
Network mask	A typical value is 255.255.255.0
IP Gateway	
DNS	Domain Name Server

All addresses are in the form n.p.q.r. Examples: 192.168.0.12, 10.0.54.123.

- ✓ *Note: in contrast to the configuration with DHCP, the “static” setting has to be reviewed each time you move the unit to a new physical site/network, as the previous IP addressing is probably not valid for the new location.*

3.5.3. Checking the IP configuration

The above configuration is kept in the unit’s memory, and reloaded at each start. It is required to restart the unit right after the initial setting, to ensure that everything is OK.

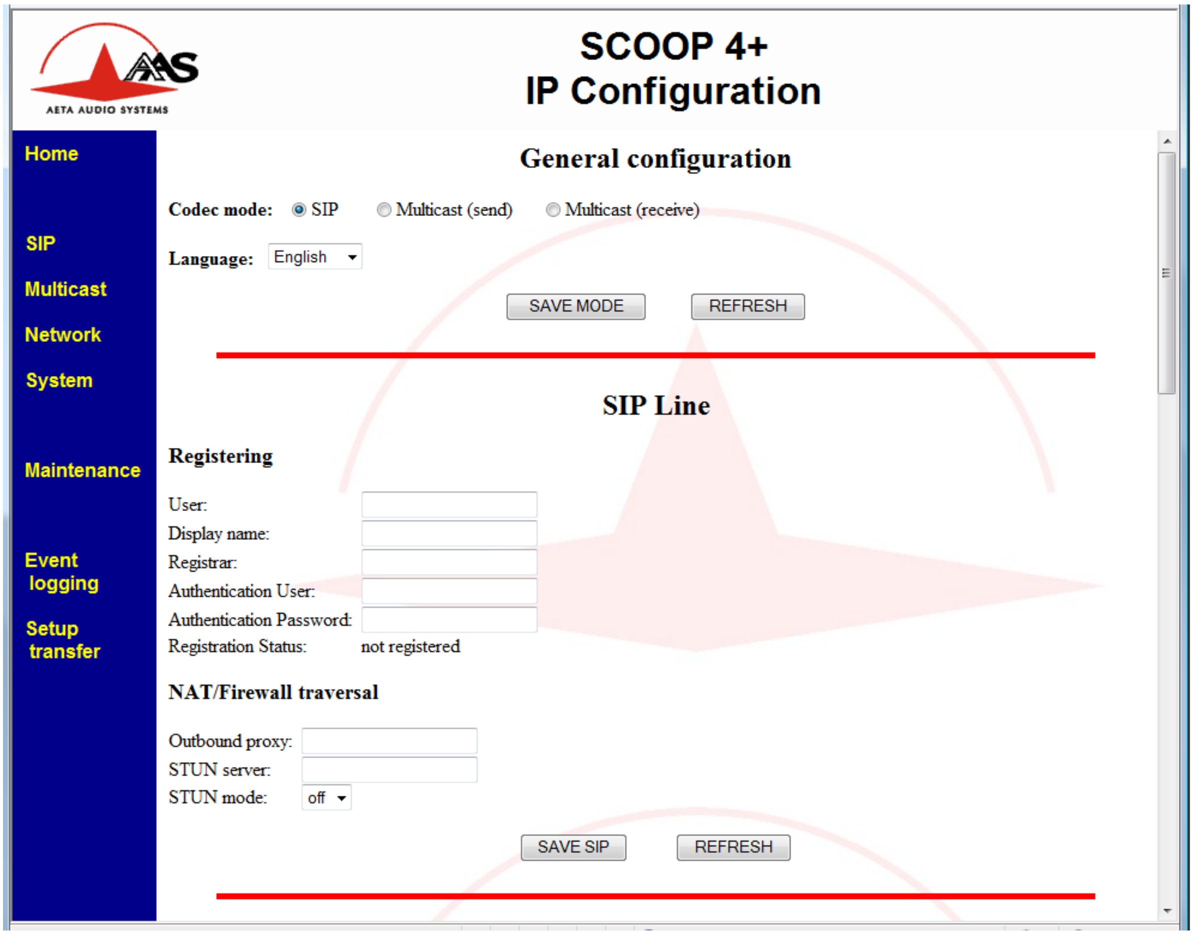
To check the setting, you can read the IP address in the “About” menu (TOOLS / Maintenance / About).

You can then also check that the unit is seen on the network and at the right address: from a computer connected to the same network, enter (in the command mode, or console mode depending on the OS) “ping *ipaddr*”, where *ipaddr* is the IP address of the SCOOP 4+.

If the response is positive, then you can proceed with the rest.

3.6. Use of the embedded html server

From a computer connected to the same network, open an html browser window and enter the IP address of the SCOOP 4+ in the “address” or “URL” field. This gives access to the html server that is embedded in the SCOOP 4+. The displayed page typically looks like the following picture:



(Note that you can select the language for this page)

If you click “Network” on the left, you can get a display of the IP addressing data. It is possible to change settings, and click the “SAVE” button[1] to write them into the SCOOP 4+. “REFRESH” reloads the page from the unit to update the display.

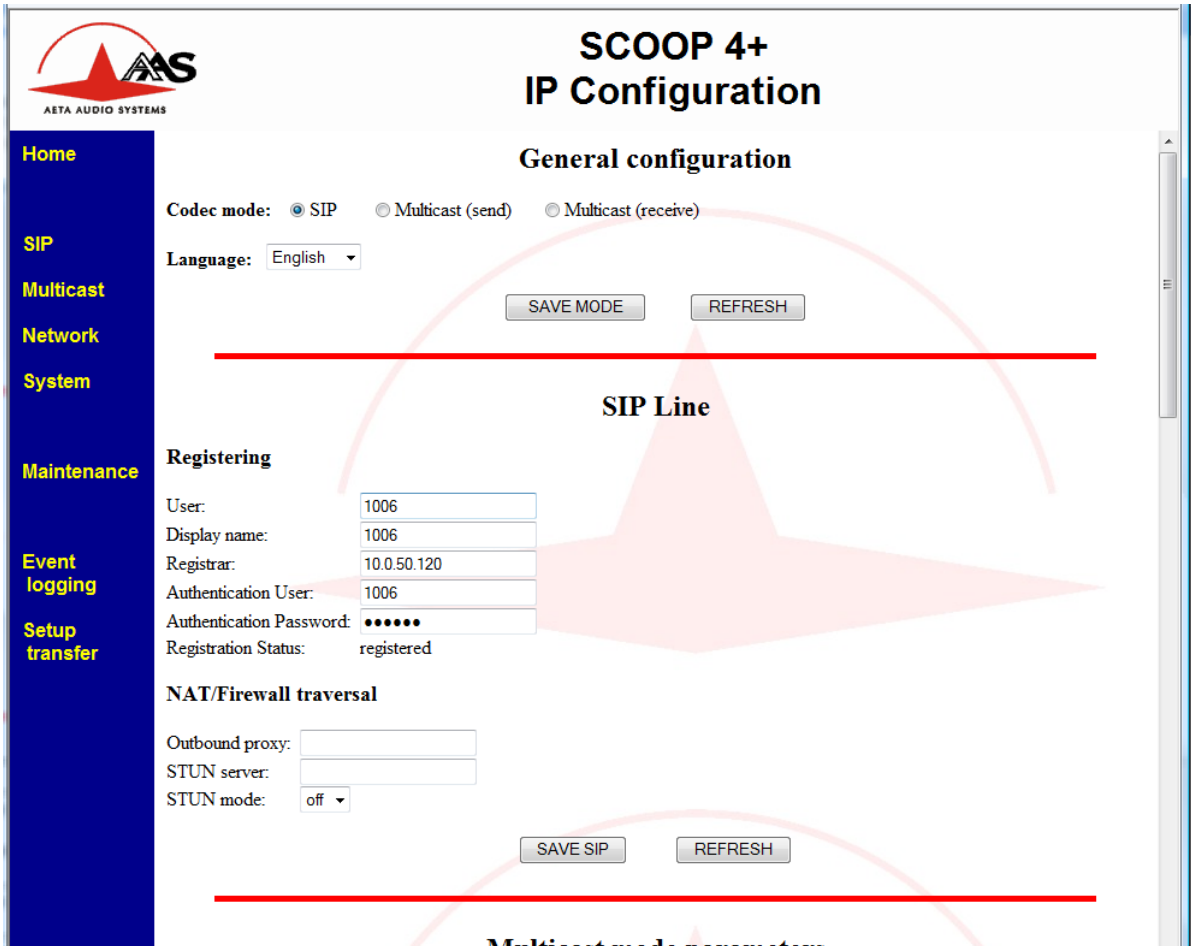
The network settings can be updated from this page, **but**:

- Obviously it is not usually possible to do the initial setting in this way!
- Be careful before changing these settings, as a wrong setting here can make you lose control over the unit… (In such event, go back to 3.5 above)

[1] Important notice : each SAVE button only uploads a section (enclosed between two bold horizontal lines), unlike the REFRESH button, which refreshes the whole page.

3.6.1. SIP registration and configuration data

You can access these parameters if you click the “SIP configuration” button on the embedded server html page. This is the only way to configure these settings, and most cannot be set from the keypad (except those mentioned in the following). The following is an example screen copy, and some comments about the displayed data:



Item	Notes
User, Display name, Authentication user	Refer to the network administrator and/or the administrator of the SIP server; Often these three parameters have the same value, as here, but they may be different.
Authentication password	Refer to the network administrator and/or the administrator of the SIP server
Registrar	IP address of the SIP registrar; a symbolic name (e.g. sipsrv.mycomp.com) is accepted, if recognised by the DNS. *Can be also read from the menu* (TOOLS / Maintenance / About)
Registration status (*read only data*)	Shows that the unit is (or is not) successfully registered on the server. *Can be also read from the menu* (TOOLS / Maintenance / About)
Outbound proxy	An outbound proxy is one way of getting access through a NAT router or a firewall; Refer to the network administrator and/or the administrator of the SIP server for this setting
STUN server	A STUN server is also one means of getting access through a NAT router. If such server is available, enter here its IP address or domain name. *Examples of usable STUN servers can be found on the "Support" page of our web site (http://www.aeta-audio.com))*
STUN mode	Enable or disable the use of the STUN server. This allows to keep the address of the STUN server even when the function is disabled. *This setting is available from the menu (SETUP / Net / Param / STUN Mode)*

Make sure to click the "SAVE SIP" button located at the bottom of this section if you want to actually write your changes into the SCOOP 4+.

✓ *For the operation in SIP (unicast) mode, make sure to leave the "Codec mode" setting (top of the page) on the "SIP" position.*

The registration data do not have to be changed often in normal operation. In fact, they may be still valid even after the unit moves to another location, even though its IP configuration changes.

3.6.2. Settings for multicast mode

To use the multicast mode, first select the appropriate mode on top of the page:

- « Multicast (send) » if the unit has to be the source of the program to be "multicast";
- « Multicast (receive) » if the unit will be receiving the multicast program.

Some complementary settings may be done for this mode. *The default settings can be suitable, but for network management issues it may be needed to force specific values.* To do that, go to the "Multicast" section of the html page.

- « Control port »; this is the UDP port (multicast) used to send the audio stream description. If this field is left blank, its default value is 6000.
- « Audio port »: this is the UDP port (multicast) used for the audio stream itself. If this field is left blank, its default value is 6001.
- « TTL » (Time To Live): leave the default value (128) unless there is a specific reason to change it.

3.6.3. “System” section

The system information includes information about the embedded software versions, the Ethernet MAC address (unique and fixed for a given unit), and the current IP address.

Be careful with the security password. This optional feature is left blank in the initial factory setting. It is **highly recommended to set a password** (non-trivial, of course) whenever the unit is installed in a non-protected environment, i.e. when non authorised persons can access the html page.

3.6.4. Maintenance section

- Firmware update: this section allows uploading updates for the embedded software. Detailed instructions are provided with the update files.
- The REBOOT button immediately restarts the unit.

3.6.5. “Event logging” section

Access is given in this section to a log recorded in the codec (internal memory card). The events are stored in text mode (unformatted ASCII), and the html page directly shows the last 50 events under the “Log” headline. It is also possible to get the complete history:

- Either by opening the full log in a separate tab or window, thanks to the “Open logfile” button;
- Or by downloading it thanks to the “Save logfile as” button.

Events are time-stamped (month, day, hour, minute, second) with the internal clock. This clock is not “real time” (no battery in the unit), but the codec is capable to synchronise at boot time with a time server using the NTP protocol. If such server is available and accessible via the Ethernet interface, enter its address in the “NTP server” field on top of this section, then click the “Save” button. A few public servers are proposed as well in the drop-down list besides “NTP server”.

✓ *Important: the date is universal time (UTC), and it does not take into account the geographic location or possible DST.*

3.6.6. “Setup transfer” section

It is possible to backup all the settings of the unit in a file (complete “snapshot”), and conversely restore a complete setup from a file previously backed up in this way.

This section allows to handle such file transfers between the codec and a computer used for html browsing:

- « Save a complete configuration file » allows to download the complete current setup of the unit in order to save it into a file on the computer.
- « Load complete configuration » allows the reciprocal process, for restoring the complete setup from a backup file up to the codec.
- « Load configuration without IP settings » fulfils almost the same function, but without acting on the IP addressing. This is often to be preferred, especially when operating the unit only from remote via IP.

✓ *Warning : files transferred in this way are devised for backup, but they are not “portable” from one firmware version to the other.*

3.7. First level maintenance

3.7.1. Internal description

To be added later

3.7.2. Internal configuration

✓ *All the configuration is done in the factory, and/or it can be changed by means of the keyboard/display interface, without having to open the unit.*

However, a few settings can be done internally by setting jumpers:

- It is possible to prevent one alarm type to light on the red alarm LED on the back of the equipment.
- It is possible to disable the standby mode (in which case the device is always in operation as long as the mains power is present).

Please consult us for such operation! We remind that unduly opening the unit can void the warranty. **In any case, opening the unit may expose live parts and is hazardous. Never open or maintain the internal parts without first disconnecting the AC supply.**

3.7.3. Analysis of malfunctions

The following table indicates the detected alarm conditions and their classification:

Alarm condition	Internal	External	Minor[1]
Power or fuse fault	X		
Bad start-up of a microprocessor, or interface fault detected on start-up	X		
Overload on an audio input			X
Fault on AES/EBU audio input		X	
Decoder synchronisation error		X	
Network clock fault[2]		X	

Table 3 - List and classification of alarm conditions

Excluding the case when an internal failure disables the management micro-controller, messages are displayed to indicate the anomaly, or the fault can be searched using the menu.

[1] Minor alarms are readable on the display, but do not trigger alarms (contacts and LEDs)

[2] Fault of the network clock source currently used for synchronisation (X21/X24 main port or secondary port)

The test loops accessible from the "TESTS" menu can help improve the analysis of a problem:

- In order to check if the audio part functions correctly, use the "Audio" loop and check if the audio is OK at the output.
- To check if the coding part functions correctly, activate "Loop 3" and check if the alarm disappears (and the decoding indicators come back to normal), and if the audio is present at the output.
- "Loop 2" sends back to the remote codec the compressed data received from the network (see the description of test loops in 2.4, Supervision and control interface, page 9). This way, it is possible to test the integrity of the transmitted data and/or check that the remote codec works properly.

The decoder out to encoder in loop ("Audio feedback" loop) can be used for overall functional check, and also for aligning the overall chain.

In leased line mode, a clock fault is one typical cause of an external alarm. This can be due to:

- complete loss of the X24/V11 interface, due to a failure of the transmission line;
- a failure of the transmission device connected to the codec;
- an incorrect clock frequency (i.e. incompatible with the codec configuration).

On the other hand, in case of a decoder alarm with no clock error, possible causes are :

- lack of signal received from the X24/V11 interface, due to a failure of the transmission device connected to the codec, or a transmission failure in the network ;
- a fault in the remote codec, or else the remote codec has an incompatible configuration ;
- transmission errors causing erratic alarms.

Errors such as "AES error" and "AES sync loss" can frequently be seen, even when the unit is configured to use the analog inputs. This is because the AES output is always active, and by default "genlocked" to the AES input. To avoid such undesired alarms:

✓ *When not using digital audio interfaces, set the digital audio sync in "Master" mode (SETUP / Audio / Digital / Synchro / Master)*

4. Detailed operating mode – User interface

In local mode, the unit is operated thanks to a keyboard and display on the front panel. The display is an alphanumeric backlit LCD with two 20-character lines.

Operating from the keyboard can be protected by a password (8 digits maximum). In such case, the password must be entered to start a session and get access to the user menus. The password can be changed or deleted by the user.

4.1. Main operation modes

There are two parameters which have a major impact on the operation of the unit and on the user interface.

First, the unit features three transmission modes: transmission over Ethernet/IP, “leased line” mode, and transmission over the ISDN.

In comparison with the permanent leased line connection, IP and ISDN modes are “dial up” modes and bring a number of additional parameters to be controlled:

- dial number and/or full SIP URI for the destination of a call;
- call set up and control;
- device SIP registration data, or local ISDN number and sub-address;
- miscellaneous network operation parameters.

The status display is slightly different in order to recall the transmission mode currently in use.

Second, in the leased line mode or ISDN mode the unit can be operated either as a normal “single codec”, or as a “dual codec” capable to transmit two independent 7 kHz bandwidth audio channels. This aspect has a big influence on the way the device is installed, set up and monitored.

In the following, the main operation modes are shortly designated as: “IP mode”, “ISDN mode” or “LL mode” (for leased line mode), and “Single codec” or “Dual codec”.

4.2. Equipment start-up

During start-up, the unit displays temporary messages. This initialisation lasts around one minute. Then the main menu is displayed.

At this stage, if the configuration includes a non-blank password, the keypad is locked and the password must be entered in order to access the menus: just enter the password (1 to 8 numbers), and the unit is unlocked as soon as the last digit is entered. On factory setting or after erasure of the unit memory, the password is blank so this step is skipped.

4.3. Description of the keyboard

The LED indicators and the main function keys are described in 3.2.1 (Front panel, LCD and basic control keypad). In its middle section, the front panel includes an alphanumeric keypad used for entering numbers and/or texts. This keypad is used in a similar way to a mobile phone:

- The keypad works in "numeric" only mode, or in "alphanumeric" mode (where both numbers and letters can be entered).
- In numeric mode, only the numbers are used, and the "*****" key (as a separation between ISDN address and sub-address).
- The "alphanumeric" mode is the default mode, automatically active whenever this makes sense (not for dialling an ISDN number). A letter can be entered by pressing repeatedly a key; for example, a "B" is entered by hitting the "**2**" key three times (sequence 2, A, B). Use the "vertical arrow" (same key as the # sign) in order to switch between capital and lower case letters. The "**1**" key is used for various symbols which are not all marked on the keypad: « **@** », « **:** », « **-** », « **_** »…
- For dialling in IP mode, it is possible to force the numeric mode (instead of the default alphanumeric mode). First press the "*****" key; the ">" prompt will appear, and then only numbers can be entered.

4.4. Description of the menus

The unit features a tree-structured menu, and the three function keys on the bottom of the LCD are used to navigate through the menus. The OK key is used to confirm some settings or enter data, and the "Esc" key allows to go back to the upper menu level. Pressing this key several times makes sure you come back to the main default menu.

From the top menu, you can directly enter one of the three main menus by hitting the function key just beyond:

- TOOLS: maintenance and housekeeping functions, and access to status information
- DIR: access to the directory
- SETUP configuration of the codec

The "TOOLS" menu is itself divided into three sub-menus:

- Status information about the status and alarms
- Maintenance test and maintenance functions
- Misc miscellaneous functions and settings

The "SETUP" menu is also divided into three sub-menus:

- Net selection and configuration of network interface and parameters
- Audio configuration of audio interfaces and parameters
- Cod selection and configuration of the coding algorithm

The following diagrams show the various sub-menus and accessible parameters.

Note that the "*" character in these diagrams shows the default and/or factory reset value for a given parameter.

4.4.1. TOOLS/About and TOOLS/Maintenance sub-menus

Notes

- **TOOLS**
  - Status
    - <Alarm 1>
    - …
    - <Alarm n>
  - Maintenance
    - About
      - IP ADDRESS — *displays IP address*
      - VERSION — *displays version info*
      - EIM — *displays version info*
      - SOFTWARE NS — *displays "software" serial number*
      - SIP REGISTRAR — *displays SIP registrar's address, if any*
      - REGISTRAR STATUS — *displays status (OK or NOK)*
    - Ethernet Setup
      - DHCP
      - Manual
        - IP Address
        - Mask
        - Gateway
        - DNS
    - User Access — *switch Full ↔ Reduced mode*
      - Confirm
    - Password — *blank after "factory reset"*
      - Enter new password
        - Confirm password
    - Tests — *
      - No test loop
      - Audio loop
      - Loop3-codec
      - Loop2-Network
      - Audio Feedback

4.4.2. TOOLS/Misc sub-menu

Notes

- TOOLS
  - Miscellaneous
    - General Reset
      - CLEAR ALL SETTINGS ?
        - Yes
        - No
    - Antenna Control — *only proposed in the specific "Inmarsat BGAN" version*
      - Off *
      - On
    - Aux. Functions — *availability depends on the current network and coding algorithm*
      - DATA CHANNEL
        - Off *
        - On
      - RELAYS
        - Off *
        - On
      - COORD. CHANNEL
        - Off *
        - On
    - LCD Backlight
      - Off
      - On *
    - LED Brightness
      - Low
      - Medium *
      - High
    - Language
      - ENGLISH *
      - FRENCH
      - ...
    - Auto Redial — *not relevant for LL mode! (but makes sense for a backup connection)*
      - Auto Redial Cod1 — *activates or not automatic redialling in case the connection is lost on codec 1*
        - Off *
        - On
      - Auto Redial Cod2 — *activates or not automatic redialling in case the connection is lost on codec 2*
        - Off *
        - On
      - Redial Attempts
        - 1 *
        - ..
        - 20
        - Infinite — *use with caution in ISDN mode!*
      - Time before dial — *time to wait before a redial attempt*
        - 1 *
        - ..
        - 30
    - Loop Control
      - Off * — *A pulse on the control loop releases a running ISDN or IP connection*
      - On — *Outgoing calls are controlled by the control loop state (not relevant for LL mode!)*
      - Backup — *Use control loop to trigger backup call (for LL mode backup)*
    - 5A System — *Concerns ISDN mode only!*
      - Off
      - On *
    - LL Backup
      - Backup Call — *Selection of network for LL backup call*
        - IP *
        - ISDN
      - Backup Receive — *Allows the unit to accept an incoming ISDN or IP call*
        - Off *
        - On

4.4.3. DIR menu

DIR			Notes
Load/Call	[Profile 1] ↕ [Profile n]		Select profile (scroll), then OK: load selected profile Green phone: load profile and call
Delete	[Profile 1] ↕ [Profile n]		Select profile (scroll), then OK: erase selected profile
Edit	[Profile 1] ↕ [Profile n]		Select profile (scroll), then OK: edit selected profile
	(OK) → ASSOCIATED COD		Edit coding algorithm for this profile
		No change	Keep existing
		Current	Replace for current setting of unit
		None	No defined coding for this profile
	Name	>_	Edit profile name
	Number1	>_	Edit (first) dial number
	Number4	>_	Edit 4th dial number (if existing/needed)
New	ASSOCIATED COD		Coding algorithm for this profile
		Current	Use current setting of unit
		None	No defined coding for this profile
	Name	>_	Enter profile name
	Number1	>_	Enter (first) dial number
	Number4	>_	Enter 4th dial number (if existing/needed)

4.4.4. SETUP/Network sub-menu

Notes

- SETUP
  - Net
    - Other [to select other network]
    - Param [for Leased Line mode]
      - Interface — *Selection of the active port in normal mode*
        - LL1 * — *Main port, labelled "X24/V11/V35"*
        - LL2 — *Secondary port, labelled "ALARM + X24/V11"*
      - Mode
        - Normal * — *Normal single codec mode*
        - DualCodec — *Select double G722 codec mode*
    - Param [for IP transmission mode]
      - Network Setup
        - DHCP *
        - Manual
          - IP Address
          - Mask
          - Gateway
          - DNS
      - Network Quality — *Selection of the preferred stability / latency compromise (see "Setting up a link in IP mode")*
        - LOW
        - MIDDLE *
        - HIGH
      - STUN Mode — *Activate/disable use of STUN*
        - OFF *
        - ON
      - Multicast — *Activate/disable Multicast mode*
        - Off *
        - Send — *Multicast active, stream "sender" mode*
        - Rcv — *Multicast active, stream "receiver" mode*
    - Param [for ISDN]
      - Protocol — *Selection of local ISDN protocol*
        - "''
        - Euro ISDN *
        - "''
      - Address — *Configuration of local line (see "Setting up a link in ISDN mode")*
        - Local Number 1
        - Sub-Address 1
      - Mode
        - Normal * — *Normal single codec mode*
        - DualCodec — *Select double G711/G722 codec mode*

4.4.5. SETUP/Audio sub-menu

Notes

- **SETUP**
  - Audio
    - Analog
      - Input Impedance
        - High *
        - 600 Ohm
      - Max Output lvl — *Defines the maximum analog level reached for a maximum level decoded signal (0 dBFS)*
        - 0 dBu
        - …
        - +16 dBu *
        - …
        - +22 dBu
      - Max Input lvl — *Defines the analog input clipping level, sending a maximum level signal to the encoder (0 dBFS)*
        - 0 dBu
        - …
        - +16 dBu *
        - …
        - +22 dBu
    - Digital
      - Synchro
        - Master
        - Genlock *
      - Sampling — *Note: valid for "Master" mode ("Genlock" mode follows AES input)*
        - 32 kHz
        - 48 kHz *
        - 96 kHz
    - Input Source
      - Analog *
      - AES
    - Level Meter /HEADROOM — *Refer to "Audio monitoring" section for this setting*
      - 0 dB
      - …
      - 12 dB *
      - …
      - 40 dB

4.4.6. SETUP/Cod sub-menu

	Notes
SETUP	
Cod	
[Previous 1]	
[Previous 2]	
[Previous 3]	
[Previous 4]	
[Previous 5]	
Other	
G711	*Not available in LL mode*
G722	
SRT	
H242	*H242: only for ISDN mode*
H221	
CELP	*Only for IP mode*
4SB ADPCM	
MONO	
STEREO	
L2 J52	*Only for ISDN mode,*
CHANNEL MODE	*MPEG L2 with J52*
MONO	
DUAL MONO	
STEREO	
JOINT STEREO	
SAMPLING RATE	
16 kHz	
24 kHz	
32 kHz	
48 kHz	
BIT RATE	
64 kb/s	
128 kb/s	
192 kb/s	
256 kb/s	
384 kb/s	
ERROR CORRECTION	
MODE 0	
MODE 1	
MODE 2	
MODE 3	
MPEG L2	
[same parameters as L2 J52]	*but max 128 kb/s in ISDN mode*
,,,	
AAC-LC / HE-AAC / HE-AAC v2	*OPTIONAL, IP and ISDN modes*
CHANNEL MODE	
MONO	
STEREO	
SAMPLING RATE	
48 kHz	
32 kHz	
BIT RATE	
64 kb/s	
128 kb/s	
TDAC	*OPTIONAL*

✓ *Important notice: limitations in bit rate depend on the transmission mode, the equipment version and the software version. This is especially true for MPEG coding.*

4.5. Handling the configuration profiles

The configuration profiles make it easier to change the configuration or set up routine calls. Each memory or "profile" includes the following fields:

- *Name* allocated to the profile
- *Number(s)*: ISDN dial number, or SIP URI, or IP address, depending on transmission mode
- Associated coding

A profile may not include a number; in such case it is used for quick and safe recall of a given coding configuration.
Conversely a profile may not include an associated coding configuration; in such case it is just like a directory entry, used for instant dialling a known destination.

The configuration profiles are managed and used via the "DIR" menu (see 4.4.3); which provides the following choice:

- *Load/Call*: after browsing the ordered list (vertical arrows), the selected item can be called directly by pressing the dial key ; otherwise, on pressing "OK" the coding configuration is loaded without launching a call.
  ☺ *by typing a letter you can directly reach this letter in the profile list*
  ☺ *the horizontal arrow allows you to look the profile parameters (name, number, coding)*
- *Delete:* of course this is for erasing a profile, after browsing the list. Press OK when the profile is selected in order to delete it.
  This deletion cannot be cancelled!
- *Edit*: after browsing and selecting the profile, it is possible to selectively edit its fields. Press OK to select the profile; then for each field, make desired changes, press OK, until all fields have been edited. The profile is then changed definitively.
- *New*: creation of a new profile. It is possible to enter or not a number, depending on the intended use for the profile. For coding, the choice left is either not to include it in the profile, or to use the current one. Of course this means the desired coding must have been configured (using the CONFIG / Cod menu) before creating the new profile.

The *Profile Manager* software eases the management of the profiles. Running under Windows, it allows more user-friendly editing of the profiles on a computer. The available features are the following:

- "Off line" profile editing on the computer thanks to the software;
- Loading the complete profile set from the computer to a Scoop4+, via an IP link (provided that the Scoop4+ is connected to the same LAN as the computer);
- Reading the profile set of a Scoop4+ into the computer, still via IP;
- Backing up the profile set into a file on the computer, for later use, e.g. for "copying" to another codec.

One can e.g., starting with a Scoop4+ with profiles created using the device menu, read these profiles into Profile Manager, edit them and create additional ones, then load this updated profile set into the Scoop4+. However, this is just one example among many ways of using Profile Manager.

4.6. Setting up a link in ISDN mode

4.6.1. Preliminary setup

The network interface must be configured depending on the local ISDN line that is used.

Protocol

First, the protocol should be set appropriately (SETUP / Net / Param / Protocol). The default setting is "Euro ISDN", also known as ETSI protocol. Change this setting if another protocol is needed in your location.

Local address

In some cases, it may be necessary to set the local address (or local ISDN number) of the line, and/or it is possible to assign a sub-address to the codec.

The local number allows "multiple subscriber numbering" or MSN. This number is usually the number remote equipment must dial to call your equipment. Configuring this number in the equipment is not mandatory if the equipment is directly connected to the public network. On the other hand, if the equipment is connected to a PABX, the number(s) are often required. The PABX may also impose a unique number for each B channel within the same BRI interface. In such a case, refer to the characteristics and configuration of the PABX.

- ✓ *Proper configuration of the local numbers is essential, and many problems in setting up links originate from mistakes or misunderstandings regarding this configuration. In doubt, leave this number blank! This is usually appropriate for public lines.*

Sub-address SA

This number differentiates several terminals connected to the same ISDN bus, which are allocated the same call number(s). Thus it can be useful in case other devices are connected with the Scoop 4+ on the same line.

Whenever a sub-address is set, the unit will only accept incoming calls specifically directed to this sub-address.

- ✓ *Most often, the best setting is to leave this blank!*

4.6.2. Call an ISDN number

To set the link, first set –if not already done- the unit in ISDN mode (menu SETUP / Net / ISDN). Then select the desired encoding format (SETUP / Cod).

- ✓ *You can get a fast setting by simply selecting one of the "recently used" configurations displayed.*

Then enter the ISDN number of the destination and press the "green phone" button.

If a sub-address is needed, after the number enter the "*" character and the sub-address (4 digits max.). The number then has the form nnnn*ssss, e.g. 0912345678*32.

When the selected encoder needs two or more B channels, the unit asks for additional numbers. If the same number is suitable, just press the "green phone" button without re-entering a number.

An error code is displayed in case of a failure of the link establishment. Refer to annex 6.4 (ISDN error causes) to find the corresponding meaning.

4.6.3. Receiving calls

When the 5A System is active, receiving calls is simple. When a call is received, the codec automatically "unhooks" and recognises the coding algorithm and protocol used, and finally set the link automatically. On the receiving side, the unit will "follow" the calling unit.

When the 5A System is not active, you should first configure the codec for the desired coding algorithm and configuration. When a call is received, the unit will synchronise with the calling device, but the link will usually fail if the calling party has used another coding configuration than expected. *However, if J52 is used by both parties, the link will succeed even without 5AS active.*

- ✓ *Important notice : the unit can receive and accept incoming ISDN calls even when it is set in IP mode. On the other hand, it will ignore such calls if it is running in LL mode, except if the"Backup receive" mode is active (for details see below 4.10, Setting up a backup link).*

4.6.4. Quick redialling

Once a number (or a couple of numbers) has been called, it is easy to recall it without having to type it again: press the "green phone" key, then you can scroll through the "history" (last dialled numbers) using the arrows. Press the phone key when the desired number is displayed. This is especially useful for quickly redialling the previous number.

4.7. Setting up a link in IP mode

A link is set up in a similar way as an ISDN link. The difference is mainly that instead of the telephone number, we use either an IP address, or a SIP URI (Uniform Resource Identifier).

4.7.1. Directly call an IP address

This is the most basic way of setting the link. It is suitable only if:

- The other unit is “directly” reachable, i.e. there is no NAT Router or firewall blocking the connectivity. The simplest case is when both units are on the same LAN.
- The IP address of the other unit is known.

To set the link, first set the unit in IP mode (menu SETUP / Net / IP), and set the desired encoding format (SETUP / Cod).

✓ *Note that you can get a fast setting by simply selecting one of the “recently used” configurations displayed.*

Then enter the IP address and press the “green phone” button.

✓ *When operating in this way, it is preferable to leave blank the SIP registering data.*

4.7.2. Calling via a SIP server

This is the technique when both units are registered on a SIP proxy server. In this case, each unit is identified by its SIP URI, in the form username@sipservername, like an email address. There is no need to know any IP address (and hence there is no problem if the IP address of a unit changes for whatever reason).

To set the link, first set the unit in IP mode (menu SETUP / Net / IP), and set the desired encoding format (embedded server, or simply from the keypad SETUP / Cod).

Then enter the SIP URI of the unit to call, and press the “green phone” button.

✓ *It is often possible to dial the short form username (omitting the @sipservername) when the device is itself registered onto the same “sipservername” server.*

4.7.3. Receiving calls

This is very simple, in both cases (direct peer to peer link or SIP server). There is nothing to do…

When a call is received, the units negotiate automatically a commonly acceptable coding algorithm, and set the link automatically. On the receiving side, the unit will “follow” the calling unit.

✓ *Important notice : the unit can receive and accept incoming IP calls even when it is set in ISDN mode. On the other hand, it will ignore such calls if it is running in LL mode, except if the”Backup receive” mode is active (for details see below 4.10, Setting up a backup link).*

4.7.4. "Network quality" setting

Depending on the quality of service provided on the network, especially its jitter performance, it is possible to change the stability/latency compromise used by the Scoop 4+. For this purpose, a setting is available in the menu (SETUP / Network / Param / Network Quality). Three choices are proposed:

- "HIGH": suitable for a good quality and low jitter network; latency is minimal, but the codec will have little tolerance to possible jitter
- "MIDDLE": intermediate (and default) setting, suitable for a moderate transmission jitter
- "LOW": to be preferred when the network has low QoS, especially for residential ADSL lines. This setting ensures a safer operation, at the cost of a high latency.

On a LAN and/or private network with a controlled quality, the "HIGH" quality setting is recommended, as it yields minimum latency.

On the contrary, it should be avoided for a link via the Internet, as it can only tolerate a low jitter. One solution can be to start with a "MIDDLE" setting, and switch to the "LOW" setting if too much audio disturbance is heard.

4.7.5. Links with IP phones

SCOOP 4+ is compatible with IP phones that use the SIP protocol (many on the market do). The algorithm used in this case is G711, but a few IP phones can also accept G722.

Note that "IP phones" include software SIP phones implemented on computers.

4.7.6. Notes about the keypad

- Use the "up arrow" key to switch between low case and capital letters.
- Once a number or SIP URI has been called, it is easy to recall it without having to type it again: press the "green phone" key, then you can scroll through the "history" (last dialled numbers) using the arrows. Press the phone key when the desired number is displayed. This is especially useful for quickly redialling the previous number or URI.

4.7.7. Multicast mode transmission

On a network that can support it[1], the multicast can optimise the resource usage when an audio stream has to be distributed simultaneously to several destinations. Contrary to the normal bidirectional unicast mode, this mode is unidirectional: one *sender* encoder sends a stream towards a multicast *group address*, and one or several decoders receiving the stream pick up the packets sent to this group address and decode the audio stream.

On the Scoop 4+ the operating mode stays quite similar to the "normal" mode, with mainly two differences in the multicast mode:

- A codec must be set as sender or receiver device
- SIP is not used and hence the SIP configuration is not relevant

For the operation it is assumed the network "statically" supports UDP multicast, i.e. routers of the network recognise and deal with routing the packets with multicast group addresses.

[1] This does not include the Internet; multicast cannot be used over the Internet.

For obvious reasons the coding setup is entirely decided at the source. On the codec on the audio source side (hence *sender* of the encoded stream), the procedure for setting up a multicast stream is the following:

- Set the codec in "multicast send" mode;
- Select on this codec the desired audio coding parameters;
- Start the streaming as for setting up a call in normal mode: enter the IP group address, then press the key.

✓ *Note that the codec decodes its own stream on its audio output; this can be useful for audio monitoring.*

For each codec that has to decode the stream, the procedure is simple:

- Set the codec in "multicast receive" mode;
- Start the decoder as for setting up a cormal call: enter the IP group address, then press the key.

4.8. Auto-redial feature

In LL mode, the unit is always supposed to be linked, and it will transmit and look for a reception signal as long as it is on. In contrast, the IP mode and the ISDN modes are "dial up modes", where a link can be set up and released at will. When it is necessary to hold the link on permanently, outgoing calls may be backed up by using the auto-redial function. When it is active, the codec can redial automatically in case a connection fails. The redial capability applies in three situations:

- If the initial call fails for any reason (e.g. called party is busy); the codec then redials and retries to establish the link.
- The codec can also redial if the link is already established and the link is lost, for any reason else than "local release" (e.g. the remote unit mistakenly dropped the line).
- After a power failure, after rebooting the codec will automatically redial and set-up the link back.

✓ *Note that, while "auto redial" is active, an established link can be definitively stopped only by releasing the line on the calling codec side.*

It is possible to program the time period that the unit will wait before redialling after a failed trial, and it is also possible to program the maximum number of times the codec will redial before giving up.

The activation of this function and the configuration of its parameters can be found in the "Auto Redial" sub-menu (TOOLS / Misc / Auto Redial menu). In dual codec mode, the function can be activated separately for each codec.

4.9. Loop control

In normal operation for IP or ISDN mode, outgoing calls are sent or released using the menus and/or the remote control interface. When the loop control function is selected, outgoing calls are controlled by activating or not optically isolated input loops. One loop is available for each codec when in double codec mode. When the input loop is activated (i.e. current is flowing), the corresponding codec establishes a link by calling the last number (or SIP URI in the IP mode) previously dialled by the unit. When the loop is de-activated, the codec releases the line and stays idle as long as the loop is not active (except if receiving an incoming call).

✓ *In normal operation, it is nevertheless possible to release a running connection by briefly activating ("pulse") the control loop.*

The "auto-redial" feature is implicitly active when loop control is active: the codec tries to keep the link, and automatically recalls the remote unit if the line drops, as long as the input loop is active. The "time before redial" parameter described in the above is also applicable to the loop control mode. On the other hand, the "redial attempts" parameter is not applicable here, because the unit will always try to recover the link, until the loop is left inactive.

✓ *Note that, as an important consequence, when using loop control, the termination of a link must always be done on the calling party side by de-activating the input loop. Whenever the line is released by the receiving party, the calling unit will redial and re-establish the link.*

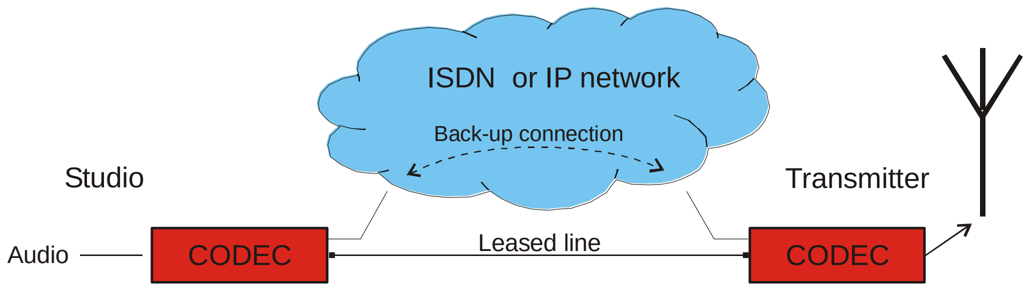
When loop control is active, the input loops are the only means of setting up an outgoing call; setting a call from the menu is not allowed.

✓ *Remind that the first step is to set-up the link once in normal mode, and later activate the loop control mode; afterwards the input loop is used to trigger a redial to the previous ISDN number or SIP URI.*

4.10. Setting up a backup link

The SCOOP 4+ has the capability for backing up a permanent leased line audio link thanks to an ISDN connection, or an IP connection.

A typical example of this application is shown in the following diagram, where a permanent leased line is used for transmitting a radio programme from a studio to a transmitter. A codec is installed on each side of the link, and each codec is also connected to the ISDN or an IP network. In this way, transmission via the ISDN (or respectively IP) can be used as a temporary back-up in case the normal permanent link fails. Once the problem on the normal line is solved, the connection on ISDN (resp. IP) can be released and normal operation in LL mode is restored.



The SCOOP 4+ makes such back-up easy because both transmission interfaces (normal leased line and backup interface, ISDN or IP) are available in the same unit. Moreover, the SCOOP 4+ helps to automate the switchover process, especially on the transmitter side, where most often quick human intervention is not possible. The following describes one way of configuring such a system, and details the resulting backup process.

4.10.1. Basic principles

When a failure of the leased line is detected, one of the two codecs has to switch to ISDN (resp. IP) "backup" mode, and then call the other unit over the backup network access. The audio transmission in then provided over the backup link. Later on, when the leased line comes back to normal, the backup connection can be released and both codecs have to switch back to LL mode and normal operation.

We designate as "caller" the codec which switches first and initiates the call over the backup network, and "receiver" the codec which switches on receiving such a backup call.

4.10.2. Caller codec: setup and operation

For the switch to backup (following a failure on the leased line), two methods can be used:

- Either the switchover is "manual": an operator on the studio side can switch the studio codec to ISDN (IP) mode and launch a call to the other site via the backup medium. For this method, no special preliminary setting is needed.
- The other method is to use the "loop control" feature as described above in section 4.9. In this case, the whole switchover sequence can be executed by simply activating the input loop. Conversely, releasing the input loop brings the unit back to normal operation in LL mode.

For this latter method, the best situation is when the leased line status is monitored by an alarm contact that closes when the line is down. This alarm can be used for activating the control loop, so that the studio codec quickly and automatically switches over to backup mode (and also comes back to normal when the leased line recovers); no manual intervention is needed in such case.

In order to use this operating mode, two settings have to be done on the “caller” codec:

- Select the network to use for backup: menu TOOLS / Miscellaneous / LL Backup / Backup Call…
- Activate the loop controlled backup: menu TOOLS / Miscellaneous / Loop Control / Backup

✓ *Important notice: when switching to backup mode the number called is the last dialled! For this reason it is important to dial and call the backup destination in the desired mode at least once, before setting the unit in its normal LL mode.*

4.10.3. Receiver codec: setup and operation

The setup is easy on the “receive” side of the backup link, as it is somewhat “passive”. The “backup receive” mode must be activated, via the menu TOOLS / Miscellaneous / LL Backup / Backup Receive.

Once this is done, when the codec is operating in LL mode, on receiving a call from the ISDN interface or the IP interface, the codec switches to the ISDN or IP mode, answers the call and the connection is established with the remote codec. Later, when the temporary connection is released (by the calling party), the “receiver” codec automatically switches back to LL mode, and normal operation is resumed.

✓ *Reminder: in LL mode the codec accepts no incoming call from its IP or ISDN interfaces, unless the backup receive mode is active.*

✓ *Conversely, once this is done, the codec accepts incoming calls regardless of the network interface, IP or ISDN.*

✓ *Thanks to the automatic coding algorithm detection (5AS in ISDN mode or SIP/SDP in IP mode), the coding setup is not critical on this “receiver” side. However, for increased safety in ISDN mode one may prefer to force the configuration (by de-activating the 5AS).*

4.10.4. Notes regarding the automatic switchover

Note that, when switching from LL to ISDN or IP mode, a separate set of parameters is recalled. As a consequence, the coding configuration can be made, if desired, totally different in the back-up ISDN or IP mode.

Of course, both units should be configured in an adequate configuration for each transmission mode, and then each codec can be set in the LL mode to start the normal operation.

✓ *For the ISDN (and backup) configuration, DO NOT set the units in double codec mode. Otherwise, the automatic switchover will not work properly.*

4.11. Erasing and resetting the configuration

In some cases like e.g. if the password is forgotten, it may be necessary to restart from the factory default setting.

To erase the entire configuration and load the factory default settings, you should normally go to the "General Reset" sub-menu (TOOLS / Misc / General Reset).

- ✓ *In case the keyboard is locked and you have lost the password, hold the "hang up" key pressed for 10 seconds; this forces the system directly to the "General reset" sub-menu. You can then confirm that you really want to erase all the settings, including the password…*

The factory default password is blank.

4.12. Backing up and recalling the configuration

It is possible to backup all the settings of the unit in a file (complete "snapshot"), and conversely restore a complete setup from a file previously backed up in this way. Such transfers are done using the embedded html server; refer above to 3.6.6 for the operation.

5. Technical characteristics

5.1. Characteristics of interfaces

5.1.1. Analogue audio inputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated.

The inputs are balanced type, using 3-pin female XLR sockets.

Maximum input level:	adjustable from 0 to +22 dBm ± 0.3 dB
Nominal input impedance:	600 Ω or 10 kΩ (menu setting, SETUP / Audio / Analog / Input Impedance)
Impedance balance:	*TBD*
Common mode rejection ratio:	> *TBD* dB (measured with Z = 600 Ω)

5.1.2. Analogue audio outputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated. The outputs are balanced type, using 3-pin male XLR sockets.

Maximum output level:	adjustable from 0 to +22 dBm ± 0.3 dB.
Nominal load impedance:	600 Ω or 10 kΩ
Output impedance:	<50 Ω
Symmetry:	> *TBD* dB (Z_L = 150 Ω)

5.1.3. Digital audio input and output

These interfaces comply with recommendations:

- AES3-1992
- EBU Tech. 3250-E
- CCIR Rec. 647

They support (in genlock mode) a sampling rate from 28 to 96 kHz. In master mode, the unit can be set in one the following sampling rates: 32, 48 and 96 kHz.

5.1.4. Headphone output (front panel)

This output (6.35 mm jack on front panel) is for the connection of a 32 Ω headphone. It is also possible to plug a high impedance headphone; however, the maximum available power will be lower.

5.1.5. Main X24/X21/V11/V35 interface

The X24/V11 interface uses a 15-pin male Sub-D connector. The following table shows the pinout.

Signal		Pin		Signal	
Frame ground		1			
Transmitted data	Ta	2	9	Tb	Transmitted data
		3	10		
Received data	Ra	4	11	Rb	Received data
Indication	Ia	5	12	Ib	Indication
Received clock	Sa	6	13	Sb	Received clock
		7	14		
Electrical ground		8	15		

The codec does not transmit a C signal.

The codec can also be connected to V35 interfaces; a specific adaptation cable is needed in such case. The connection is described in Annex 6.5, (V35 interface adaptation).

5.1.6. "Alarm + X24/X21" interface

This interface uses a 15-pin male Sub-D connector. The following table shows the pinout.

Signal		Pin		Signal	
Internal alarm - Common	**IA-Com**	1			
Transmitted data	Ta	2	9	Tb	Transmitted data
Internal alarm - NC	**IA-C**	3	10	**IA-O**	**Internal alarm - NO**
Received data	Ra	4	11	Rb	Received data
Indication	Ia	5	12	Ib	Indication
Received clock	Sa	6	13	Sb	Received clock
External alarm - NO	**EA-O**	7	14	**EA-C**	**External alarm - NC**
Electrical ground		8	15	**EA-Com**	**External alarm - Common**

The bold text refers to the alarm contacts. Both are form-C type. The "NO" terminal is open when the alarm is set, otherwise it is connected to the "Common" terminal. The "NC" terminal is connected to the "Common" terminal when the alarm is set, otherwise it is open.

The current and voltage handling capabilities of the relays (static type) are:

Maximum output current: 120 mA

Maximum output voltage: 350 V peak

Resistance of output loop: < 35 Ω

Isolation > 1500 V_{RMS}

The codec does not transmit a C signal.

The codec can also be connected to V35 interfaces; a specific adaptation cable is needed in such case. The connection is described in Annex 6.5, (V35 interface adaptation).

5.1.7. Remote control interface

This interface uses a 9-pin female Sub-D connector on the rear panel. This is a V24/RS-232 type interface with only Tx and Rx signals (no flow control). The following table indicates its pinout (DCE type pinout).

Pin		Function	
2	Rx	V24 data to the PC	Output
3	Tx	V24 control data, from the PC	Input
5		Ground	
Other		Not connected	

The interface is configured as follows: 4800 bauds, 8 bits, no parity, one stop bit.

5.1.8. Data interface (« data »)

This V24 interface uses a 9-pin female Sub-D connector on the rear panel. Like for the remote control interface, only Tx and Rx are used, there is no flow control, and the pinout is of DCE type.

Pin		Function	
2	Rx	Received V24 data	Output
3	Tx	Transmitted V24 data	Input
5		Ground	
Other		Not connected	

The data interface is configured as follows: 8 bits, no parity, one stop bit. It is possible (see menu TOOLS / Misc / Aux. Functions / DATA CHANNEL) to activate the interface and to configure its baud rate (300 to 9600 bauds). However, the maximum allowed baud rate depends on the audio coding used (see 2.6.1 - Data channel).

5.1.9. Ethernet interface

This RJ45 socket has standard Ethernet pinout (for use of a normal “straight” cable to an Ethernet hub or switch).

The installation and operation of this function is detailed in 3.5, Initial setup of the Ethernet interface.

5.1.10. “Digital I/O” interface

TBD

5.1.11. Relay transmission interface ("AUX" socket)

The relay transmission interface (refer to 2.6.2, Relay transmission) is available on the 25 pin female sub-D "AUX" Socket. It includes two isolated current loop inputs and two dry contact outputs.

The following table shows the pinout of the socket for this function:

Pin	Function
13	Output loop n°2 (b)
25	Output loop n°2 (a)
12	Output loop n°1 (b)
24	Output loop n°1 (a)
11	Input loop n°1 (b)
23	Input loop n°1 (a)
10	Input loop n°2 (b)
22	Input loop n°2 (a)
9	+5V of internally supplied power supply
21	0V of power supply

All loops are isolated and bi-directional (free polarity).

The characteristics of the input loops are:

Input loop control current: 6 mA (max. 100 mA)

Resistance of input loop: ~ 560 Ω (current limiting series resistor)

Input loop isolation: > 1500 V_{RMS}

A +5V to +12V source may be connected directly on an input loop, because the internal series resistor is dimensioned for this purpose. For a higher voltage source, it may be necessary to limit the input current.

The characteristics of the output loops are:

Maximum switching voltage (output): 350 V peak

Maximum switching current (output): 120 mA

Resistance of output loop: < 35 Ω

Output loop isolation: > 1500 V_{RMS}

The 5V power supply is available from the unit to power a low-consumption device (maximum 300 mA current consumption), or e.g. to power the input loops or LED indicators connected to the output loops.

5.1.12. Coordination channel interface (“AUX” socket)

In addition to the loop control and relay transmission interfaces, the (optional) coordination channel input and output are available on the 25-pin female sub-D connector (“AUX” Socket on the rear panel), with pinout as indicated hereunder.

The input and output are balanced floating signals, transformer isolated.

Maximum level: 9 dBm

Impedance: 600 Ω

Nominal bandwidth: 300 – 3400 Hz

Pin	Function
1	Coordination channel output (-)
14	Coordination channel output (+)
2	Frame ground
15	Coordination channel input (+)
3	Coordination channel input (-)
16	Frame ground

5.1.13. Loop control interface (“AUX” socket)

The 25 pin female sub-D “Aux.” socket includes isolated current loop inputs and dry contact outputs, that can be used to remotely control the calls and indicate the link status:

- The input loops have an effect only if the “loop control” function is enabled (see 4.9, Loop control). The output loops are always operative.
- Activating the input loop #1 triggers an ISDN or IP call on the codec (codec 1 only if the unit is configured as a double ISDN codec); de-activating the loop releases the line.
- Activating the input loop #2 triggers an ISDN call on codec 2 if the unit is configured as a double ISDN codec; de-activating the loop releases the line. This loop has no action in single codec mode.
- Output loop #1 is closed while an ISDN or IP connection is running, or while codec 1 is linked if the unit is configured as a double ISDN codec ;
- Output loop #2 is closed while an ISDN connection is running on codec 2, if the unit is configured as a double ISDN codec ;

The following table shows the wiring of the socket for this function:

Pin	Function
17	Input loop n°2 (a)
5	Input loop n°2 (b)
18	Input loop n°1 (a)
6	Input loop n°1 (b)
19	Output loop n°2 (a)
7	Output loop n°2 (b)
20	Output loop n°1 (a)
8	Output loop n°1 (b)
21	0V of power supply
9	+5V of internally supplied power supply

All loops are isolated and bi-directional (free polarity).

The characteristics of the input loops are:

Input loop control current:	6 mA	(max. 100 mA)
Resistance of input loop:	~ 560 Ω	(current limiting series resistor)
Input loop isolation:	> 1500 V_{RMS}	

A +5V to +12V source may be connected directly on an input loop, because the internal series resistor is dimensioned for this purpose. For a higher voltage source, it may be necessary to limit the input current.

The characteristics of the output loops are:

Maximum switching voltage (output):	350 V peak
Maximum switching current (output):	120 mA
Resistance of output loop:	< 35 Ω
Output loop isolation:	> 1500 V_{RMS}

The 5V power supply is available from the unit to power a low-consumption device (maximum 300 mA current consumption), or e.g. to power the input loops or LED indicators connected to the output loops.

5.1.14. Connector for DC input (*option*)

The "12V DC" version of the product includes, in place of the AC power socket, a 4-pin male XLR socket for the connection of an external DC source.

The ground is connected on pin 1, and the +12V source must be supplied on pin 4.

5.2. Audio performance

The audio performance in this part applies to the system without coding/decoding, and excluding the coordination channel. The additional effect of the audio encoding and decoding on audio performance depends on the coding algorithm used and its parameters.

Except when differently stated, the following measurements are done at a +6 dBm input level and on the AD/DA path, with maximum input and output level set at +16 dBu.

5.2.1. Transmission gain

The drift in time of the gain from the input to the output of the codec is less than ± 0.3 dB.

5.2.2. Amplitude-frequency response

All measurements are done with a +6 dBm input signal, and a reference frequency of 1020 Hz. The measurements are done with a loopback before coding/decoding, so the possible effect of compression has no influence.

To be detailed

5.2.3. Group delay distortion

Taking the minimum group delay as reference, the group delay distortion on the AD/DA path is always less than 1 ms.

5.2.4. Idle channel noise

Background noise is measured with no audio modulation (idle channel), with maximum input and output level set at +16 dBu, through the whole encoder-decoder chain (wide band coding, with 48 or 32 kHz coding frequency).

Maximum noise level[1]:	- 56 dBm
(quasi-peak detection, CCIR weighting)	(or - 62 dBq0ps)

This result in a signal to noise ratio (SNR) of more than 72 dB.

When the maximum input and output level is set at another level, both the signal and noise levels are shifted but the SNR remains in the same range.

5.2.5. Total distortion vs. frequency and level

Total distortion relative to maximum level (or THD + N) is less than –82 dB over the whole audio bandwidth (20 – 20 000 Hz). This performance holds for audio signals from –80 dB to –1 dB relative to the maximum level (+16 dBu).

5.2.6. Crosstalk

Crosstalk is less than -80 dB over the whole bandwidth.

5.2.7. Gain and phase difference between channels

The gain difference between channels is less than ± 0.3 dB over the whole bandwidth, for any sampling frequency.

The phase difference between channels is less than ± 3 degrees over the whole bandwidth, for any sampling frequency.

[1] Worst case for all types of algorithms; MPEG performs better than the others

5.3. Network protocols and ports

The Scoop 4+ implements or complies with the following protocols (non exhaustive list):

- Physical and link layers: Ethernet, 100BaseT, 10BaseT
- Network/Transport layers (IPv4) : TCP/IP, UDP/IP, RTP/IP
- Application : HTTP, Telnet, DHCP, STUN
- Audio transmission: SIP signalling, SDP, RTP, RTCP, RFC3550/3551, RFC3640
- Compliant with EBU recommendation Tech 3326 (interoperability of audio codecs for contribution)

The ports used by the device are the following:

Type	Port	Désignation	Notes
TCP	80	HTTP	Embedded html server
	6000	Control	Remote control (« command line » mode) ; used by Scoop4Man and TeleScoop
UDP	3478	STUN	For querying a STUN server (optional)
	5004	RTP	For audio streaming
	5005	RTCP	For audio streaming
	5060	SIP	SIP signalling
	6000	Multicast/description	Multicast stream description channel
	6001	Multicast/audio	Multicast stream transmission channel

The ports for multicast mode can be changed from the default values in the above table.

5.4. Power supply

The codec operates from mains 85-263Vac, 47-63 Hz. Protection is provided by a resettable fuse. The maximum power consumption is 15 W (without the coordination channel option).

A product version is available which is powered from a +12V DC source (non isolated). The unit includes protection against overcurrent (resettable fuse on the DC input) and overvoltage. The DC source voltage must be in the 10V-15V range. The maximum consumption is 1 A (for a 12V DC source).

5.5. Dimensions and weight

The unit is a 19 inches frame of 1U height (44 mm or 1.75”) and 252 mm depth (12.5”).

Its weight is about 3.2 kg.

5.6. Environmental characteristics

The equipment operates over a 0°C to 45°C ambient temperature range (32°F to 113°F), and a 5% to 90% humidity ratio range.

The SCOOP 4+ complies with “CE” directives regarding safety and EMC.

- Safety: compliance with EN60950
- EMI: radiated emissions complying with EN55103-1
- Susceptibility: compliance with EN55103-2

5.7. Versions - Options

The various versions available for the SCOOP 4+ are the following:

- SCOOP 4+ DUO LL/IP: with Ethernet/IP and leased line interfaces
- SCOOP 4+ TRIO LL/IP/ISDN 2B: with Ethernet/IP, leased line interfaces and one ISDN interface
- SCOOP 4+ TRIO LL/IP/ISDN 4B: with Ethernet/IP, leased line interfaces and two ISDN interfaces

Besides, certain functions are available as options:

- MPEG AAC-LC audio coding;
- MPEG HE-AAC and HE-AAC v2 audio coding;
- Audio coordination channel;
- TDAC algorithm for ISDN mode;
- 12V DC power (instead of AC mains)

5.8. Accessories and related products

The SCOOP 4+ is delivered with a mains cord and a CAT5 Ethernet cable.

Along with the coordination channel option, a specific cable is delivered, which provides XLR plugs for the coordination channel input and output (input on a female plug, output on a male plug).

For remote controlling SCOOP 4+ units from a PC, the Scoop4Man configuration and supervision software is available (for a Windows computer).

The TeleScoop™ supervision software is also available and allows to control the other rackmount codecs of the Hifiscoop, Scoop 3 and Hifiscoop 3 ranges (please consult us).

6. Annexes

6.1. Complements on the algorithms and protocols used

6.1.1. Auxiliary data in the MPEG frames

The auxiliary data are used for the following purposes:

- Reed-Solomon error detection and correction (J52 standard)
- Data channel
- Other auxiliary information: relay transmission, and/or auxiliary audio channel. The insertion of this auxiliary information is an extension (AAS proprietary format) to MPEG. However, the frame structure remains compliant.

6.1.2. Reed-Solomon encoding

In order to cope with possible transmission errors in the network, Reed-Solomon error correction coding can be added, compliant with J52 recommendation. Four correction modes are available in the SCOOP 4+:

- Mode 0 : no error correction, Reed-Solomon coding disabled
- Mode 1 : protection of only the control information and scale factors in the MPEG frame, low redundancy (so-called “unequal protection”)
- Mode 2 : protection of the whole frame, moderate (2.5 %) redundancy (so-called “low equal protection”)
- Mode 3 : protection of the whole frame, high (10 %) redundancy (so-called “high equal protection”)

Higher redundancy increases the protection against errors, but slightly degrades the audio quality, as redundancy takes up part of the bit rate that could be allocated to audio coding.

Most often, for a normal quality transmission link, mode 1 is sufficient and it consumes little bit rate from the compressed data, so it hardly impacts the audio quality. Although J52 does not apply to leased line connections, this error protection technique is also implemented in the SCOOP 4+ in leased line mode.

6.1.3. H221 framing

H221 defines a framing structure that allows byte synchronisation recovery in leased lines, and the transmission of control data along with the main data.

6.1.4. Proprietary coding algorithms

The “proprietary” coding algorithms are algorithms which are not standardised by the ITU-T but have distinctive features that make them useful for some applications:

- Low coding-decoding delay (4SB ADPCM);
- Low bit rate wide band speech coding (CELP)
- TDAC (Time Domain Aliasing Cancellation, MDCT-based algorithm)

6.2. Overview of the SIP protocol

6.2.1. What is SIP?

SIP is for Session Initiation Protocol, a protocol specified by the IETF for establishing media transmission sessions. SIP is considered the communication protocol of the future by most vendors, and as such it has deep influence on the VoIP applications.

As a signalling protocol, SIP brings methods and techniques to solve the issues related to the establishing of an audio link. Almost as important, it is a recognised standard, implemented on many network devices and systems. Using SIP helps you build modular and really evolutive systems, not being tied to a single vendor.

The SIP protocol is an essential requirement in the EBU Tech 3326 recommendation (a.k.a. “N/ACIP” recommendation).

6.2.2. Setting a link with SIP

Let us look at an example (diagram below): a reporter on the move with a Scoopy+[1] wants to make a call to a SIP compliant codec located in the main station. The reporter may be at home, or at another location, not necessarily known in advance.

Once the Scoopy+ is on and connected to the network, it will register itself ❶ to a SIP “registrar”. This registrar can be located on the LAN of the radio house, but it may as well be elsewhere in the network. Then the registrar “knows” where the Scoopy+ is, what its IP address is. On the radio house side, a similar process takes place ❷.

To call the codec of the radio house (e.g. a SCOOP 4+), the reporter just needs to know its SIP address, which can be like studio12cod@radiomcr.com (indeed very similar to an e-mail address). To call the unit, the reporter has to select the preferred audio coding mode on the Scoopy+ (e.g. mono G722), then call the remote unit, simply using this SIP address (SIP URI).

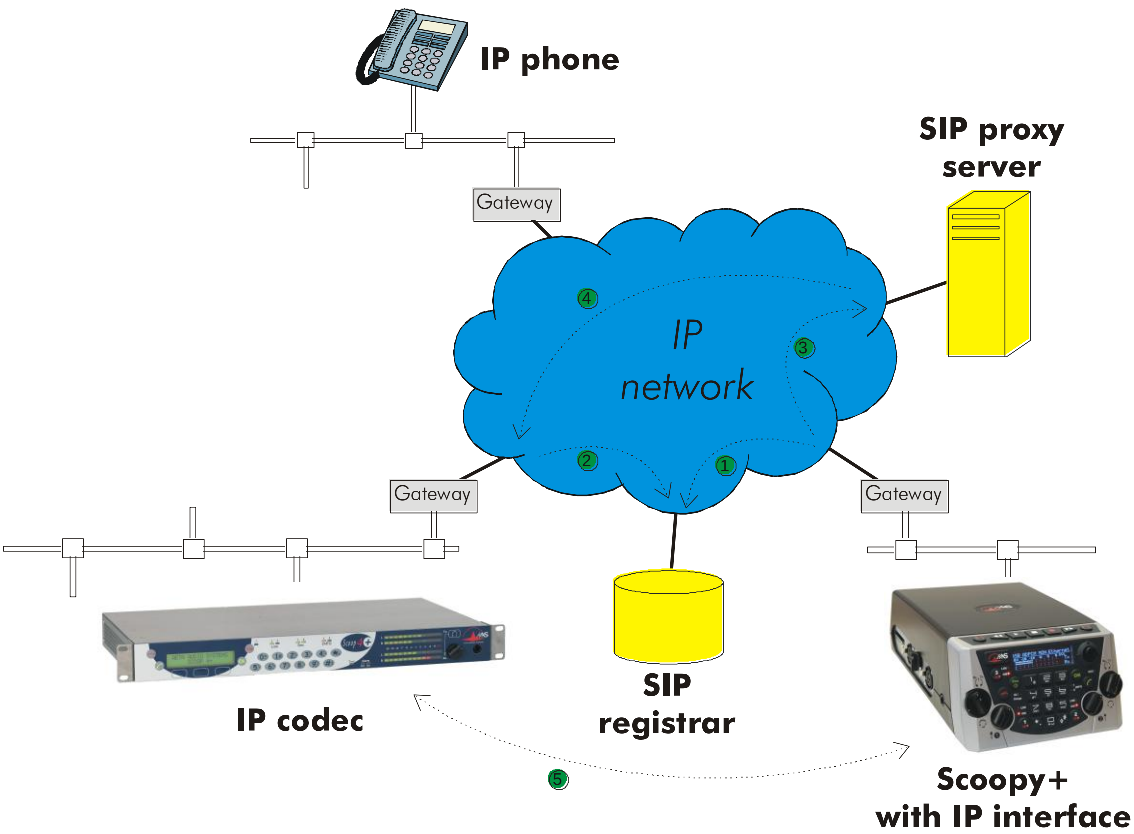
What happens then on the network: the Scoopy+ sends the request ❸ (INVITE in SIP protocol) to a proxy server (often the same device is also the registrar). To make things simple, this proxy then relays and routes the request ❹ to its destination. Resolving the SIP URI to a physical network and address uses mechanisms similar to those used for resolving URLs. Several proxys may be involved in cascade to finally reach the desired destination, but this does not have to be known or dealt with by the end devices. The following is like the initiation of a phone call: the IP codec “rings”; if it accepts the call, this is notified to the Scoopy+.

At this stage, the proxy(s) provide the Scoopy+ and the IP codec with all the address data they need for the link, then the actual audio streams can be exchanged ❺ between both units. As a very important feature, the end devices now can exchange data directly; the proxys do not have to be on the path, they are only involved in the setting (and later the ending!) of the session. The codecs will automatically exchange their coding capabilities, and agree on a coding mode, with no further user intervention.

Alternatively, the call can be done from the station to the reporter, in a way very similar to the above. In contrast with ISDN links, the operators at the station do not even need to know where the reporter is located! This is because the registrar deals with this issue.

Note that it is also possible to set a link with a SIP-compliant VoIP phone instead of another codec. This is one of the numerous advantages of using a standard.

[1] Scoopy+ is a portable audio codec from AETA AUDIO SYSTEMS; the description here applies to both Scoopy+ and the Scoop 4+, as they are both SIP compliant and mutually compatible


IP phone
SIP proxy
server
Gateway
IP
network
4
3
2
1
Gateway
Gateway
SIP
registrar
IP codec
Scoopy+
with IP interface
5

6.3. Some methods to deal with NAT routers and firewalls

Problems arise when the desired connection has to go through a NAT router, and/or a firewall, that blocks a direct IP communication.

This is a very common issue, especially if one needs to set up a transfer via the Internet. It is impossible here to describe in details the possible ways to deal with this problem, but the following just shortly discusses some typical solutions. Most probably, a network administrator should be consulted for support, and for granting adequate network authorisations and/or privileges.

The most classical issues are related to:

- The presence of a NAT router on the network path between the codecs.
- The presence of a firewall on this path.

It is always important to have available the information regarding the network organization and to be allowed to access the devices which need to be configured. Hence we highly recommend to involve the persons empowered for such tasks.

6.3.1. Links via a private network

No special problem should be met within a LAN. The operation is also possible with codecs from other manufacturers, provided that they comply with the Tech3326 EBU recommendation (also known as "N/ACIP" recommendation). However you should check for specific settings or preparation possibly needed on such devices.

A wide area network covers a wider geographic range, and the network topology most probably includes routers on the path between the codecs to be linked. However, usually there is not much difference with a local area network.

Note: using a VPN leads to just the same case; the operation is identical as far as the codecs are concerned.

6.3.2. Links through a public network (Internet)

If each of the units has got a "direct" access to Internet with a public address, we are in the same situation as the previous one, functionally speaking (private WAN). The addressing scheme is normally static, as DHCP can rarely be used on a public access. In fact, this situation is very seldom met in the field!

First, the Internet access is usually protected by a firewall which will, as a principle, block a priori the desired connection. In such case exceptions (to the firewall security rules) must be created, that will allow this connection; this has to be done by the person in charge of the network management.

Most often, on one access if not both, the codec accesses the Internet via a NAT router. This router shares Internet access, with one or a few public addresses, among the equipment on the LAN. On this LAN the devices get local private addresses, and the router carries out an IP address translation. Note that:

- As an example, a consumer ADSL modem-router is almost always a NAT router, sharing a single public IP address between the devices connected to the router.
- It is just the same on a 3G/3G+ mobile IP access ; the terminals (phones or computers) access the Internet via NAT routing.
- NAT routing is often included in the firewall features; in fact NAT routing somewhat participates to the protection against direct attacks from the outside.

NAT routing is an obstacle to transmission with UDP, mainly for two reasons:

- It does not allow unsolicited data to come in from the outside network. In other words, data input is accepted on a port as an answer to a request from the local network, but an external agent cannot directly initiate the transmission of a packet.
- The terminal units on the LAN only know their private local address. On the other hand, agents implementing the SIP protocol have to communicate to each other the addresses and ports to be used for the media exchanges. Because of the NAT routing, agents do not get the real public addresses, which leads to failure of the session setup attempts.

We are now looking at various methods used to overcome these obstacles.

NAT and use of the STUN server

The STUN protocol is a method which is often successful[1] in helping the agents to discover their public address even when they are « hidden » behind a NAT router. Here is the operation principle:

- A STUN server is used, which is accessible over the Internet ;
- The address of this server is programmed into the agent (i.e. the audio codec in our topic);
- The agent queries the server and discovers its public IP address and port number, as seen from outside of the NAT router and LAN;
- This addressing information is then used by the agent for negotiating and setting up a media session.

The STUN server address is programmable in the html page of a Scoop 4+ or Scoopy+. Besides, one can also find in the menu (keypad and display on the front of the unit) an activate/disable (on/off) selection, available without having to clear the server address.

The are many public STUN servers available on the Internet; here are a few examples, valid at the time of writing:

Domain name	**Numeric address**
stun.xten.com	75.101.138.128
stun.ekiga.net	75.101.138.128
stun.sipgate.net	217.10.79.2
stun.ippi.com	208.73.210.27
stun.sipphone.com	198.65.166.165

Examples of STUN servers

It is advisable to check that the server is operative. A list of servers can also be found on the support page of our web site http://www.aeta-audio.com.

[1] Although not with so called « symmetric » NAT routers

Standard NAT router

Situation: codec A behind a NAT router with no specific programming (a codec accessing Internet via a mobile network is almost always in such situation). We also assume that the other codec (B) is accessible with a public address.

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B cannot call codec A

Advantages	**Drawbacks**
Configuration is relatively simple	B cannot call A
No change is needed on the router	
Several codecs can be set behind the NAT router	
Method suitable for mobile network access	

NAT router with DMZ

Situation: codec A behind a NAT router and placed in « DMZ ».

We also assume that the other codec (B) is accessible with a public address.

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B can call codec A, using the public address of the NAT router

Advantages	**Drawbacks**
Each codec can set up a session	Need to configure the router
A is nearly equivalent to a codec with a direct public access	Only one codec can be set up in this way on a LAN
	A is exposed to external attacks
	The DMZ may be already reserved for other network equipment
	Method not possible for a mobile network access

NAT router with port forwarding

Situation: codec A behind a NAT router and configuration of the router to forward to A the necessary ports.
We also assume that the other codec (B) is accessible with a public address.

Port forwarding to be set on the router:

- UDP 5060 (=SIP port)
- UDP 5004 (RTP port) and 5005 (RTCP port)[1]

[1] For Scoop4+ versions before 1.32, ports 9000 and 9001 respectively

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B can call codec A, using the public address of the NAT router

Advantages	**Drawbacks**
Each codec can set up a session	Need to configure the router
A is nearly equivalent to a codec with a direct public access	Only one codec can be set up in this way on a LAN
	Method not possible for a mobile network access

Use of a SIP server

In addition to the numerous features it brings along, using a SIP proxy server is a very powerful method to solve the issues related to NAT routers, because most SIP proxies are capable to detect the presence of NAT routers and/or deal appropriately with their traversal.

If a SIP server is available, and once the codecs are registered on this server:

- Any registered codec can call another registered[1] codec, regardless whether there is or not a NAT router on the path.
- The identifier (SIP URI) is stable and does not depend on the location of the called agent ("mobility" feature).

It is possible either to use a public server on the Internet, or to install a private server accessible via the Internet.

Advantages	**Drawbacks**
Each codec can initiate a session Each codec can receive calls	Installation may not be easy (private server)
Identification is simple and location/time-wise stable	Reliability of server questionable (public server)
Security : a private proxy can be linked with a firewall	
Also works with symmetric NAT routers	
Interoperation with telephony over IP	
Method suitable for mobile network access	

[1] Depending on the access control policy, a server may accept « outgoing » calls to third party domains, or accept « incoming » calls from non registered agents.

6.3.3. Summary and reminder of essential rules

The table below sums up the situations where a link can be set up (not using a SIP proxy server) and reminds the needed specific settings :

	Codec A access	Possible calls	Codec B access	Notes
1	LAN	⇒ ⇐	LAN (same)	
2	Private WAN	⇒ ⇐	Private WAN	
3	Internet direct	⇒ ⇐	Internet direct	
4	NAT	⇒	Internet direct	STUN needed for A
5	NAT + DMZ	⇒ ⇐	Internet direct	STUN needed for A
6	NAT + port forwarding	⇒ ⇐	Internet direct	STUN needed for A UDP ports 5004, 5005, 5060
7	NAT	⇒	NAT + DMZ	STUN needed for A and B
8	NAT + DMZ	⇒ ⇐	NAT + DMZ	STUN needed for A and B
9	NAT + port forwarding	⇒ ⇐	NAT + DMZ	STUN needed for A and B UDP ports 5004, 5005, 5060
10	NAT	⇒	NAT + forwarding	STUN needed for A and B
11	NAT + DMZ	⇒ ⇐	NAT + forwarding	STUN needed for A and B
12	NAT + port forwarding	⇒ ⇐	NAT + forwarding	STUN needed for A and B UDP ports 5004, 5005, 5060

✓ *Basic rule: Codec behind a NAT router => use a STUN server.*

This allows the codec to set up outgoing calls. This is not sufficient to be accessible to connection requests from the outside.

✓ *Mobile network access without SIP server or VPN => use a STUN server*

✓ *NAT + DMZ or NAT + forwarding => incoming calls are possible.*

Incoming calls are not possible behind a NAT router without either such change or a SIP proxy.

✓ *SIP server => maximum versatility, at the expense of some initial effort (for installation)*

Important note: contrary to a common but wrong belief, the SIP protocol (always used by AAS codecs) *does not impose the use of a SIP* ***server***. Codecs can set up point-to-point links using this protocol in the above described conditions. When no SIP *registrar* is involved, the identifiers are simply the IP addresses of the codecs.

6.4. ISDN error causes

The following table lists the call clearing causes. The error message is typically CLEARED: hh (dd), where hh is an hexadecimal number and dd its decimal value. The message meaning is given for an ETSI ISDN. Causes with values greater than 80 hex are generated internally.

Code	Meaning
01 (1)	unallocated (unassigned) number
02 (2)	no route to specified transit network
03 (3)	no route to destination
06 (6)	channel unacceptable
07 (8)	call awarded and being delivered in an established channel
10 (16)	normal call clearing
11 (17)	user busy
12 (18)	no user responding
13 (19)	no answer from user (user alerted)
15 (21)	call rejected
16 (22)	number changed
1A (26)	non-selected user clearing
1B (27)	destination out of order
1C (28)	invalid number format
1D (29)	facility rejected
1E (30)	response to STATUS ENQUIRY
1F (31)	normal, unspecified
22 (34)	no circuit/channel available
26 (38)	network out of order
29 (41)	temporary failure
2A (42)	switching equipment congestion
2B (43)	access information discarded
2C (44)	requested circuit/channel not available
2F (47)	resources unavailable, unspecified
31 (49)	quality of service unavailable
32 (50)	requested facility not subscribed
39 (57)	bearer capability not authorized
3A (58)	bearer capability not presently available
3F (63)	service or option not available, unspecified
41 (65)	bearer capability not implemented
42 (66)	channel type not implemented
45 (69)	requested facility not implemented
46 (70)	only restricted digital information bearer capability is available
4F (79)	service or option not implemented, unspecified

(continued on next page)

Code	Meaning
51 (81)	invalid call reference value
52 (82)	identified channel does not exist
53 (83)	a suspended call exists, but this call identity does not
54 (84)	call identity in use
55 (85)	no call suspended
56 (86)	call having the requested call identity has been cleared
58 (88)	incompatible destination
5B (91)	invalid transit network selection
5F (95)	invalid message, unspecified
60 (96)	mandatory information element is missing
61 (97)	message type non-existent or not implemented
62 (98)	message not compatible with call state or message type non-existent or not implemented
63 (99)	information element non-existent or not implemented
64 (100)	invalid information element contents
65 (101)	message not compatible with call state
66 (102)	recovery on timer expiry
6F (111)	protocol error, unspecified
7F (127)	interworking, unspecified
91 (145)	no signaling data link establishment
A2 (162)	no line activation
FF (255)	call clearing, unspecified

6.5. V35 interface adaptation

This annex indicates the proper connections to use when interfacing the SCOOP 4+ codec to DCE equipment using a V35 interface.

6.5.1. Connection table

The three leftmost columns show the pin allocation on the 15-pin connector of the codec.

The two columns on the right indicate the pinout on a 34-pin V35 connector or a 37-pin sub-D connector. Consult the DCE documentation for other connector types.

Only the bold indicated signals need be connected; leave other pins unconnected. However, the frame ground (pin 1) may be used for connecting the shield of the connection cord.

X24 signals	Pin	Signal direction	Function	V35 signal		Pin number 34-pin connector	37-pin connector
G	**8**		**Signal ground**	**102**	**SG**	**B**	**19**
	15						
	7						
	14						
Sa	**6**	←	**Data clock**	**115a**	**RETA**	**V**	**8**
Sb	**13**	←	**Data clock**	**115b**	**RETB**	**X**	**26**
Ia	5	←					
Ib	12	←					
Ra	**4**	←	**Received data**	**104a**	**RDA**	**R**	**6**
Rb	**11**	←	**Received data**	**104b**	**RDB**	**T**	**24**
	3						
	10						
Ta	**2**	→	**Transmitted data**	**103a**	**TDA**	**P**	**4**
Tb	**9**	→	**Transmitted data**	**103b**	**TDB**	**S**	**22**
	1		Frame ground		FG	A	1

6.5.2. Optional connections

The I signals (pins 5 and 12) may be used for signalling the codec whether the link is operative (especially, if the data clock is valid).

This is useful only in the “double codec” mode, when using both data interfaces. The codec uses this signal to detect a transmission fault on the interface, and search for synchronisation on the other interface. *In the single codec mode, the state of the I signal has no effect on the codec operation.*

To use this option, the Ia signal (pin 5) must be connected to ground (pin 8), and the signal from the DCE must be connected to Ib (pin 12). The signal from the DCE (e.g. RLSD, pin F on 34-pin connector, pin 13 on 37-pin connector) must be active as long as the connection is OK, and de-activated in case of a transmission loss.

6.6. Notice regarding open source code

The software of this product includes programs and libraries that are covered by the GNU General Public License (or "GPL"), available for example at following address: http://www.gnu.org/licenses/gpl.txt . Under this license, the source code for concerned elements is available on our Internet site ("Download" page); otherwise it can be obtained on request by e-mailing AETA AUDIO SYSTEMS (open_source@aeta-audio.com).

The software also includes the sipX stack covered by the LGPL license.

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